

UNITED STATES PATENT AND TRADEMARK OFFICE

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BEFORE THE PATENT TRIAL AND APPEAL BOARD

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SAMSUNG ELECTRONICS CO., LTD., SAMSUNG ELECTRONICS,  
AMERICA, INC., and HARMAN INTERNATIONAL INDUSTRIES, INC.

Petitioners

v.

STATON TECHIYA, LLC,

Patent Owner

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Case IPR2024-01015

U.S. Patent No. 11,665,493

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**DECLARATION OF CHRIS KYRIAKAKIS, PH.D.  
IN SUPPORT OF PETITION FOR *INTER PARTES* REVIEW  
OF U.S. PATENT NO. 11,665,493**

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I, Chris Kyriakakis, declare that:

**I. ASSIGNMENT**

1. I have been retained by Samsung Electronics Co., Ltd., Samsung Electronics America, Inc. (together “Samsung”), and Harman International Industries, Inc. (“Harman”) (collectively “Petitioners”) as an independent expert consultant in this proceeding before the United States Patent and Trademark Office (“PTO”).

2. I am being compensated at a rate of \$525/hour for my services in this proceeding, which is my regular and customary rate.

3. My compensation is in no way contingent on the nature of my findings, the presentation of my findings in testimony, or the outcome of this or any other proceeding. I have no other interest in this proceeding.

4. I have been asked to consider whether certain references disclose or suggest the features recited in the claims of U.S. Patent No. 11,665,493 (“the ’493 patent”) (Ex. 1001).<sup>1</sup> My opinions are set forth below.

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<sup>1</sup> Where appropriate, I refer to exhibits I understand will be attached to the petition for *inter partes* review of the ’493 patent (the “Petition”).

## **II. BACKGROUND AND QUALIFICATIONS**

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

6. I am over 18 years of age and, if I am called upon to do so, I would be competent to testify as to the matters set forth herein. A copy of my current curriculum vitae, which details my education and professional and academic experience, as well as a list of all publications I have authored in the past ten years, is included as Ex. 1003 in this proceeding. The following provides an overview of some of my experience that is relevant to the matters set forth in this declaration.

### **A. Relevant Academic Experience**

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

8. Since 2002, I have been a tenured Associate Professor in the Electrical Engineering Department at USC. My research interests lie at the intersection of

acoustics, psychoacoustics (the science that studies human perception of sound), and audio signal processing. My recent research has focused on the study of audio systems in challenging environments including automobiles and mobile devices, as well as algorithms for enhancing the performance of voice recognition engines. I have published nearly 100 technical papers, including several peer reviewed papers. I have published a book entitled *Immersive Audio Signal Processing*, and hold several patents in acoustic measurement of loudspeakers in rooms and cars, loudspeaker crossover optimization, and loudspeaker response correction using signal processing. My publications examine various aspects of sound measurement, how sound interacts with the acoustical elements of the environment, novel methods for surround sound recording and reproduction, and the perception of sound by human listeners.

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

reflections, absorption, and diffusion in the performance of loudspeakers. I have also published several technical papers on acoustical measurement methods in rooms and device enclosures and developed novel signal processing algorithms for optimizing sound system performance. Other topics I have researched include multichannel audio acquisition and rendering, virtual microphones and virtual speakers, hybrid headphone-loudspeaker rendering methods, and advanced signal processing techniques for optimizing sound quality from small portable devices.

**B. Relevant Professional Experience**

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

11. In 2019, I joined Syng as the Chief Audio Scientist. Syng creates innovative products for next generation spatial audio and my role there includes research and development of audio algorithms for sound capture using microphone arrays to measure the acoustics of the room and loudspeaker beamforming for rendering sound in various directions.

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

13. I also developed audio algorithms and designed speakers. These speakers were novel acoustical designs that used a combination of unique enclosures, and audio signal processing to optimize their performance and overcome limitations that arise from small drivers and enclosures. For example, we used signal processing technologies combined with novel acoustical design to extend the bass response of small woofers and passive radiators beyond what was previously possible in small speaker enclosures. The innovations in these designs have received awards, including Popular Science's "Best of What's New."

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

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

16. In the late 1990s and early 2000s, I was a faculty researcher and later Deputy Director of the National Science Foundation's engineering research center established at USC. I was studying the fundamental and technological limitations of immersive audio and the role of acoustics on the performance of loudspeakers and audio systems in homes and cars. In 2003, together with one of my graduate students, I received the award for Best Paper at the Institute of Electrical and Electronics Engineers ("IEEE") Conference on Signals, Systems and Computers.

### **III. MATERIALS REVIEWED**

17. The opinions contained in this declaration are based on the documents I reviewed, my professional judgment, as well as my education, experience, and knowledge regarding electrical engineering, computer science, and audio and acoustic sciences.

18. In forming my opinions expressed in this declaration, I reviewed the following materials:

- the '493 patent (Ex. 1001);
- the file history of the '493 patent (Ex. 1004);
- U.S. Provisional Application 61/098,250 (Ex. 1005);
- U.S. Patent Application 12/555,864 (Ex. 1006);
- U.S. App. Pub. No. 2007/0299362 ("*Epley*") (Ex. 1007);
- U.S. App. Pub. No. 2005/0096899 ("*Padhi*") (Ex. 1008);

- U.S. App. Pub. No. 2005/0123146 (“*Voix*”) (Ex. 1009);
- Certified Translation of Japanese Patent Application Publication 2002-204500 (“*Hayashi*”) (Ex. 1010);
- Japanese Patent Application Publication 2002-204500 (Ex. 1011);
- U.S. Patent 5,182,774 (“*Bourk*”) (Ex. 1012);
- U.S. Patent App. Pub. No. 2008/0107287 (“*Beard*”) (Ex. 1013);
- U.S. Patent No. 10,979,836 (“’836 patent”) *Ex Parte* Reexamination, Final Rejection (07/28/2023) (Ex. 1014);
- U.S. Patent No. 6,567,524 (“*Svean*”) (Ex. 1015);
- U.S. Patent No. 7,039,195B1 (Ex. 1016);
- *Staton Techiya, LLC et al v. Samsung Electronics Co., Ltd. et al.*, EDTX-2-21-cv-00413, Dkt. No. 285, Claim Construction Order U.S. Patent No. 7,039,195B1 (U.S. Patent No. 9,609,424 (“the ’424 patent”)) (Ex. 1017);
- Patent Owner appeal - ’836 EPR (Ex. 1018);
- U.S. Patent No. 6,671,643 (Ex. 1019);
- any other materials I refer to in this declaration in support of my opinions.

19. All of the opinions contained in this declaration are based on the documents I reviewed and my knowledge and professional judgment. My opinions have also been guided by my appreciation of how a person of ordinary skill in the

art would have understood the claims and the specification of the '493 patent at the time of the alleged invention, which I have been asked to initially consider as no earlier than 9/19/2008 (the filing date of U.S. Provisional Patent Application 61/098,250). My opinions reflect how one of ordinary skill in the art would have understood the '493 patent, the prior art to the patent, and the state of the art at the time of the alleged invention.

20. Based on my experience and expertise, it is my opinion that certain references disclose and/or suggest all the features recited in claims 1-20 ("challenged claims") of the '493 patent, as I discuss in detail below.

#### **IV. LEVEL OF ORDINARY SKILL IN THE ART**

21. I have been informed and understand that, in the context of an invalidity analysis, a person having ordinary skill in the art is a hypothetical person who looks to prior art at the time of the invention. I further understand that the factors that may be considered in determining the level of ordinary skill include: (1) the problems encountered in the art; (2) the prior art solutions to the problems encountered in the art; (3) the rapidity of innovation; (4) the sophistication of the technology; and (5) the education level of active workers in the field. I understand that these factors need not all be considered for the analysis and that one or more of these factors may control.

22. I was asked to provide my opinion on the level of one of ordinary skill in the art with respect to the alleged invention of the '493 patent as of 9/19/2008. Based on my consideration of the factors above, I believe a person having ordinary skill in the art would have had a bachelor's degree in electrical engineering, computer science, audio engineering or a similar field and at least two years of experience with hearing instruments such as earphones, hearing aids and/or hearing protectors. The person having ordinary skill in the art could have also obtained similar knowledge and experience through other means.

23. As of 9/19/2008, I met, and in fact exceeded, the qualifications of a person of ordinary skill in the art. To be clear, all of my opinions in this declaration are from the perspective of one of ordinary skill in the art as I have defined it here during the relevant timeframe.

## **V. RELEVANT LEGAL STANDARDS**

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

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

26. Petitioners' counsel have informed me that references or products that fall into one or more of these categories are called "prior art," and that to anticipate a patent claim a single reference must contain all of the elements and limitations recited in the claim either expressly or inherently.

27. Petitioners' counsel has informed me that for the prior art to inherently disclose a claimed limitation, the prior art need not expressly disclose the limitation, so long as the claimed limitation necessarily flows from a disclosure in the prior art.

28. Petitioners' counsel has informed me that a patent claim can be considered to have been obvious to a person of ordinary skill in the art at the time the application was filed in view of the prior art. This means that, even if all of the requirements of a claim are not found in a single prior art reference, the claim is not patentable if the differences between the subject matter in the prior art and the subject matter in the claim would have been obvious to a person of ordinary skill in the art at the relevant time, which I have been informed in this case is 9/19/2008.

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

- the level of ordinary skill in the art at the time the application was filed;

- the scope and content of the prior art; and
- what differences, if any, existed between the claimed invention and the prior art.

30. Petitioners' counsel has informed me that a single prior art reference can render a patent claim obvious if any differences between that reference and the claims would have been obvious to a person of ordinary skill in the art.

Alternatively, the teachings of two or more references may be combined in the same way as disclosed in the claims, if such a combination would have been obvious to one having ordinary skill in the art. In determining whether a combination based on either a single reference or multiple references would have been obvious, it is appropriate to consider, among other factors:

- whether the teachings of the prior art references disclose known concepts combined in familiar ways, and when combined, would yield predictable results;
- whether a person of ordinary skill in the art could implement a predictable variation, and would see the benefit of doing so;
- whether the claimed elements represent one of a limited number of known design choices, and would have a reasonable expectation of success by those skilled in the art;

- whether a person of ordinary skill would have recognized a reason to combine known elements in the manner described in the claim;
- whether there is some teaching or suggestion in the prior art to make the modification or combination of elements claimed in the patent; and
- whether the innovation applies a known technique that had been used to improve a similar device or method in a similar way.

31. Petitioners' counsel has informed me that one having ordinary skill in the art has ordinary creativity and is not an automaton. Petitioners' counsel has also informed me that in considering obviousness, it is important not to determine obviousness using the benefit of hindsight derived from the patent being considered.

32. 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).

33. I have been informed and understand that certain claim terms can be means-plus-function terms. I understand that a “means-plus-function” term is one that is expressed as a means or step for performing a specified function without the recital of structure, material, or acts in support of that function in the claim. I understand that a term is presumed not to be a means-plus-function term if it is not drafted using the term means. I also understand that a term that connotes sufficient structure to a person of ordinary skill in the art would not be a means-plus-function term.

34. I understand that for means-plus-function terms, the term is construed to cover the corresponding structure, material, or acts described in the specification and equivalents thereof. I understand equivalence may be established if the prior art element performs the same function specified in the claim in substantially the same way and produces substantially the same result.

35. I also understand that insubstantial differences between a prior art element and a claimed element—as understood in light of its specification—will not preclude a finding of invalidity. For example, a difference in a number of parts in a prior art element and the corresponding element disclosed in the patent specification may be an insubstantial difference. Similarly, if a prior art element lacks several components of an overall structure corresponding to a claimed function, that difference may be insubstantial. And potential advantages of a

structure that do not relate to a claimed function should not be considered in an equivalents determination.

36. I understand that the structures, materials, and acts to be considered equivalent or non-equivalent may be described in a patent's specification. A specification may therefore establish a narrow or broad range of equivalents, although an applicant also need not exhaustively describe the full range of equivalents.

37. If a term is determined to be a means-plus-function claim, I understand that the disclosed structure in the specification is deemed to be corresponding structure if the specification or prosecution history clearly links or associates that structure to the claimed function and the disclosure must be sufficient to achieve the claimed function.

## **VI. SUMMARY OF OPINIONS**

38. This Declaration explains the conclusions I have formed based on my analysis. These conclusions are summarized below:

- Ground 1: *Epley* in view of *Padhi* and *Voix* renders obvious claims 1, 3-5, 7-8, 10-11, 18-20.
- Ground 2: *Epley* in view of *Padhi*, *Voix*, and *Bourk* render obvious claims 9 and 12.

- Ground 3: *Epley* in view of *Padhi*, *Voix*, and *Hayashi* render obvious claim 2.
- Ground 4: *Beard* in view of *Epley*, *Padhi* and *Voix* renders obvious claims 1, 4, 6, and 13-14.
- Ground 5: *Beard* in view of *Epley*, *Padhi*, *Voix*, and *Hayashi* renders obvious claims 15-17.

## VII. OVERVIEW OF THE '493 PATENT

### A. Disclosure of the '493 Patent

39. The '493 patent, titled "ACOUSTIC SEALING ANALYSIS SYSTEM," is generally directed to "testing the seal of an orifice-inserted device." '493 patent, 1:24-27. According to the '493 patent:

[i]t can be difficult to communicate using an earpiece or earphone device in the presence of high-level background sounds. In many earpiece designs a transducer is placed near the ear canal opening. Ambient sound from the surrounding environment enters the ear canal with the audio content from the transducer. Environmental sounds such as traffic, construction, and nearby conversations can degrade the quality of the audio content.

*Id.*, 1:31-38. Additionally, the '493 patent explains that:

Thus, one method to prevent ambient sound from entering the ear is to seal or provide an acoustic barrier at the opening of the ear canal. Sealing minimizes ambient sound leakage into the ear canal, and

under the correct conditions can provide a level of noise suppression under high background noise conditions.

*Id.*, 1:41-46. Additionally, the '493 patent explains that:

[U]ser conditions may change substantially during the operation of the earpiece, and in some circumstances, the earpiece may become misaligned or may be fit incorrectly such that it is not sealed correctly. A method of seal detection is needed to optimize performance.

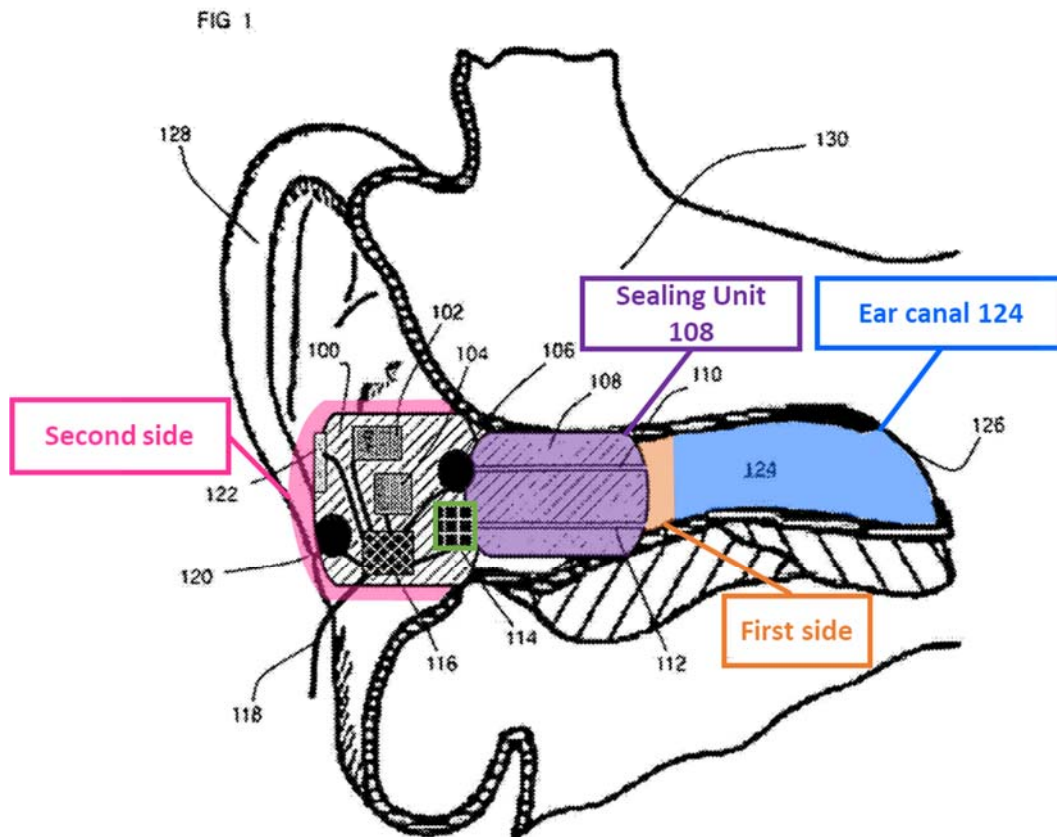
*Id.*, 1:50-55.

40. To remedy the purported problems in the art, the '493 patent proposes:

A device or a method using the device includes a balloon configured to seal a user's orifice, where the balloon is configured to produce an acoustic seal between a first side and a second side of the balloon in an ear canal. At least a second side of the balloon is fitted into the ear canal. Audio processing circuitry produces an audio signal for driving a speaker in the device and to measure sound level using output from the microphone in the device while the speaker is being driven by the audio signal. The device or method further includes control circuitry to evaluate a seal quality of the device.

*Id.*, Abstract.

41. Figure 1, reproduced and annotated below, shows an earpiece comprising **sealing unit 108** “having a **first side** corresponding to **ear canal 124** and a **second side** corresponding to the ambient environment.”<sup>2</sup> *Id.*, 4:34-36.



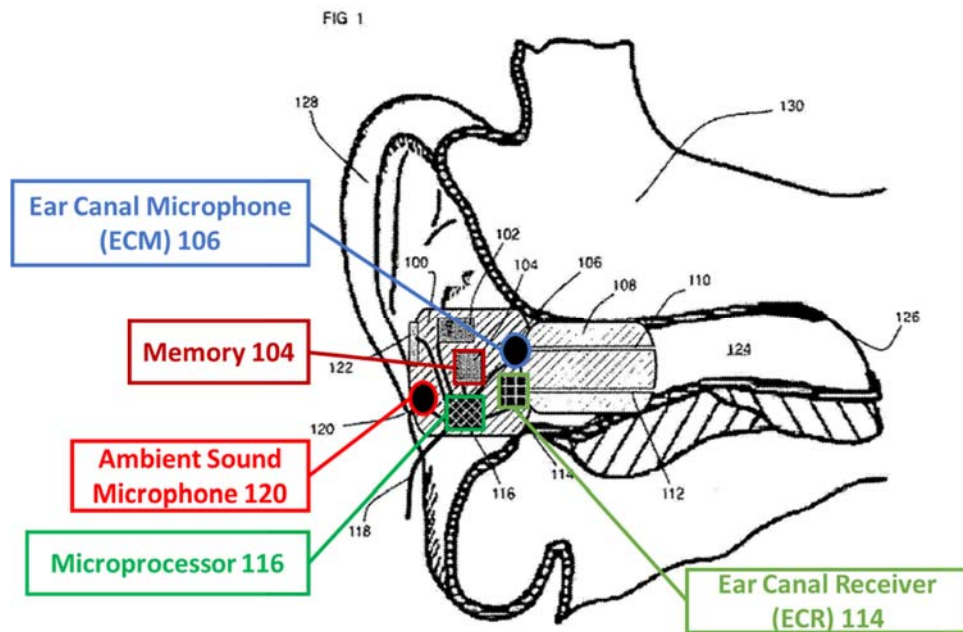
'493 patent, Figure 1.

42. The earpiece includes “an **Ambient Sound Microphone (ASM) 120** to capture ambient sound, an **Ear Canal Receiver (ECR) 114** to deliver audio to an **ear canal 124**, and an **Ear Canal Microphone (ECM) 106** to capture and

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<sup>2</sup> All emphases are added in this Declaration unless otherwise noted.

assess a sound exposure level within the **ear canal 124**.” *Id.*, 4:19-23. These components may be housed in an electronic housing unit 100, which may contain **microprocessor 116** operatively coupled to these components and to **memory 104**.



'493 patent, Figure 1.

The electronic housing unit 100 can include various system components such as a **microprocessor 116**, **memory 104**, battery 102, **ECM 106**, **ASM 120**, **ECR 114**, and user interface 122, or these components can reside in a separate system or interface operatively connected. **Microprocessor 116 (or processor 116)** can be a logic circuit, a digital signal processor, controller, or the like for performing calculations and operations for the earpiece. **Microprocessor 116** is operatively coupled to **memory 104**, **ECM 106**, **ASM 120**, **ECR 114**, and user interface 122.

*Id.*, 4:55-64.

43. In my opinion, the audio and audio processing components of the earpiece shown in Figure 1, including ones described in the above quoted sections, were standard and well-known in the art at the time of the purported inventions.

44. The '493 patent discloses exemplary techniques for seal detection, for example, with respect to Figs. 3-4, and 14.<sup>3</sup> *Id.*, 7:54-8:40 (Fig. 3); *id.*, 9:14-52 (Fig. 4); *id.*, 15:58-17:4 (Fig. 14).

45. Fig. 3 discloses determining the degree of acoustic sealing via cross-correlation of a test signal with an **ECM signal**.

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<sup>3</sup> Figure 14 and the corresponding description first appeared in U.S. Patent Application 12/555,864 filed on September 9, 2009. EX1006.

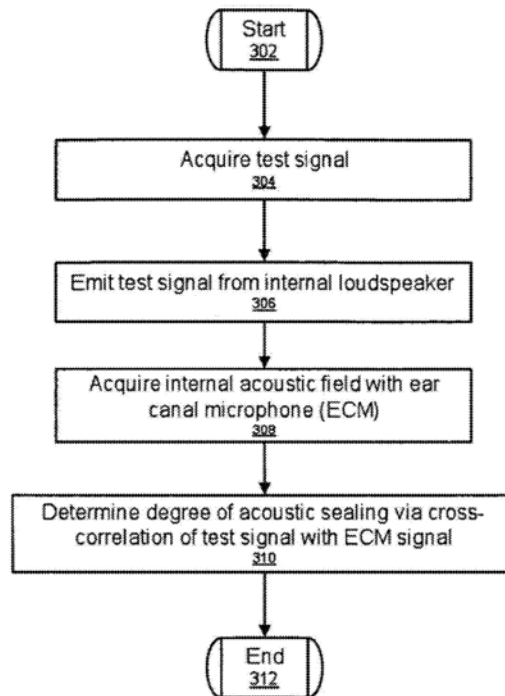


FIG. 3

'493 patent, Fig. 3.

Referring to FIG. 3, a flowchart illustrates a method for an acoustic sealing analysis system in accordance with an exemplary embodiment. In general, a first volume is acoustically isolated from a second volume. The test determines if the two volumes have sufficient acoustic isolation from one another. For example, cars are designed to have a quiet interior. Users of an automobile do not want to be subjected to the noise of the external environment. Thus, a car interior (first volume) is acoustically isolated from the external environment outside of the automobile. Similarly, an earpiece having a sealing unit such as described in FIG. 1 will create a first volume (the ear canal) that is acoustically isolated from the ambient environment of the user

(second volume). In either example, the acoustic sealing analysis system determines if there is sufficient acoustic isolation for the application. In the earpiece example, random or periodic testing of the seal may be beneficial because a new seal is formed in the ear canal when the device is put in the ear or it may shift over time depending on user activity.

The method begins at step 302. A test signal is acquired in a step 304. For example, the test signal can be stored in memory or generated by a microprocessor. The test signal is provided to the acoustic transducer. The acoustic transducer or loudspeaker (such as an ECR) emits an acoustic signal corresponding to the test signal within the first volume in a step 306. The acoustic field in the first volume is detected by an Ear Canal Microphone (ECM) in a step 308. The acoustic loading on both the ECR and ECM will change depending on the degree of acoustic sealing, thereby affecting the degree magnitude of the radiated ECR signal detected by the ECM. In general, as the degree of ear seal decreases, the effect of lumped air mass coupled to the ECR and ECM will decrease thereby increasing in Thevenin capacitance, which effectively reduces the transfer of low frequency emitted sound from the ECR to the ECM.

In one exemplary embodiment, the test signal and the acoustic signal emitted by the loudspeaker into the first volume is a single frequency sine wave signal for testing leakage from one volume to another.

The degree of sealing between the first and second acoustic volumes is determined in a step 310 and the process ends at step 312. The

cross-correlation between the emitted test signal and detected ECM signal is taken. In at least one exemplary embodiment, the test signal and the measured acoustic signal emitted by the loudspeaker are conditioned using a time delay and frequency dependent filter. The ear-seal is determined to be low (or “leaky”) if the cross-correlated signals are below a predetermined value. In at least one exemplary embodiment, automatic adjustments to the sealing section are made (such as deflating and re-inflating the sealing balloon to reseal the sealing section including retesting). Alternately, an audible sound, vocal response, or visual response can be provided to let the user know that the earpiece is sealed correctly or incorrectly.

*Id.*, 7:53-8:40.

46. Fig. 4 discloses cross-correlating a test signal and an **ECM signal** and comparing to a threshold to determine whether the seal test result should be FAIL or PASS.

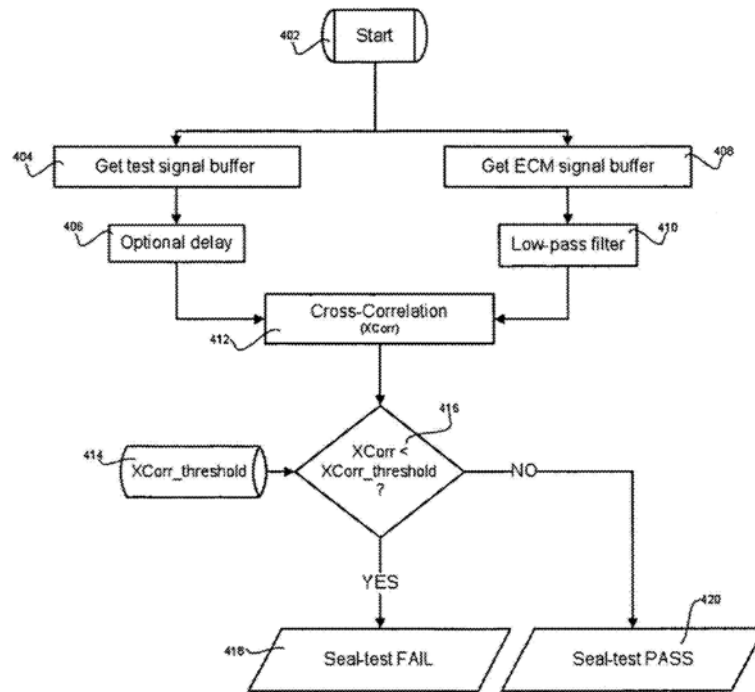


FIG. 4

'493 patent, Fig. 4.

Referring to FIG. 4 , a flowchart of an exemplary method to determine the acoustic seal integrity of an earpiece in accordance with an exemplary embodiment is illustrated. In at least one embodiment of an acoustic sealing analysis system, the test signal is masked or used in a manner undetectable by the user. This allows unobtrusive (periodic or non-periodic) testing to determine if a device is sealed correctly ensuring optimum system performance and more importantly user safety.

In at least one exemplary embodiment, an audio content is provided in a step 402. A step 404 stores the test signal in a test signal data buffer. For example, a single frequency sine wave is stored in the test signal data buffer. The output (or alternatively-input) of the test signal data

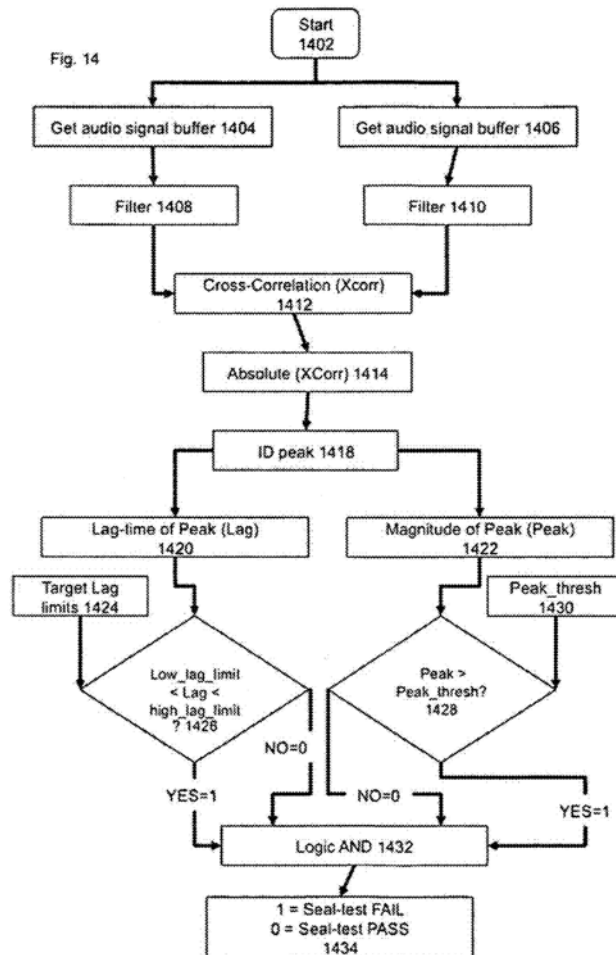
buffer is optionally delayed by digital delay unit 406. The function of delay unit 406 is to time-align the emitted test signal with the ECM signal so the cross-correlation is sensitive to changes in ear seal.

A step 408 stores the test ECM signal in a test signal data buffer. The output (or alternatively-input) of the ECM signal buffer can be filtered with a low-pass filter 410. The low pass filter can be configured so that the pass-band covers the frequency of the test signal. In one exemplary configuration, the low-pass filter can be a cascaded bi-quad IIR type filter with the cut-off frequency equal to 10 Hz greater than the test signal frequency.

A step 412 cross-correlates the optionally delayed test signal buffer with the low-pass filtered ECM signal buffer. An exemplary method for the cross-correlation algorithm is described in FIG. 5 . The instantaneous cross-correlation (i.e. the cross-correlation at zero-lag) value from step 412 is compared with the cross-correlation threshold value 414 using comparator unit 416. If the instantaneous cross-correlation of the two signal buffers is less than the threshold value 414, then the seal test status is set to FAIL 418 (i.e. an ear-seal leak is detected); otherwise, if the cross-correlation is suitably high, the seal test status is set to PASS 420.

*Id.*, 9:14-52.

47. Fig. 14 discloses another technique that involves cross-correlation of signals and comparing to a threshold to determine whether there is a proper seal.



'493 patent, Fig. 14.

Referring to FIG. 14 , a flowchart of an exemplary method to determine acoustic seal integrity is illustrated. For example, an earpiece seal integrity corresponds to a full or partial acoustic barrier between a first volume (ear canal) and a second volume (ambient environment). In one exemplary embodiment, the degree of acoustic seal integrity is expressed as either a PASS or FAIL status, where FAIL indicates that the acoustic seal is compromised relative to a normal operating acoustic seal. For example, an earpiece that has performed the seal test and determined that the sealing unit is not

sealed correctly in the ear canal of the user can provide a signal or message indicating the problem. The user can then remove, reinsert, and retest the earpiece to ensure that the seal is within normal operating specifications.

The method begins at step 1402. An acoustic test signal is provided in a first volume. In a step 1404 a transducer measures the acoustic test signal and stores it in a signal buffer. In a step 1406, a second transducer in a second volume isolated from the first volume by an acoustic barrier measures a second acoustic signal in the second volume. A portion of the acoustic test signal passes the acoustic barrier into the second volume. The amount of the acoustic test signal passing the acoustic barrier is a measure of the seal provided by the acoustic barrier.

In a filter step 1408, the measured acoustic test signal in the first volume is filtered in a frequency range corresponding to the acoustic test signal to remove signals that are not part of the test. The measured signal from the first volume is heretofore called the first volume signal. Similarly, in a step 1410, the measured signal in the second volume is filtered in a frequency range corresponding to the acoustic test signal to remove signals not related to the test (outside the frequency range) in the second volume. The measured signal from the second volume is heretofore called the second volume signal.

A correlation, cross-correlation, or coherence analysis is performed on the first volume signal and the second volume signal. The correlation, cross-correlation, or coherence analysis is a measure of the similarity

of the signals in the first and second volumes. In particular, the non-difference analysis measures the acoustic test signal leaking past the acoustic barrier by identifying the portion of the second volume signal that is similar to the acoustic test signal in the first volume.

In at least one exemplary embodiment, a correlation step 1412 is performed comprising a cross-correlation of the first volume signal and the second volume signal. In a step 1414, the peak of the cross-correlation is identified. The peak of the cross-correlation is Absolute(XCorr). In a mathematical step 1418, the Lag-time of Peak 1420 and the Magnitude of Peak 1422 is calculated. The Lag-time of Peak 1420 is a measure of the time delay between receiving the signals in the first and second volumes. In particular, the first volume signal should be received before the second volume signal. The Magnitude of Peak 1422 corresponds to the similarity between the signals in the first and second volumes. Thus, a larger number for Magnitude of Peak relates to more leakage of the acoustic test signal getting past the acoustic barrier.

Two comparisons are performed that determine if the acoustic barrier is sealed correctly based on the measured and calculated data from the first and second volumes. In a comparison step 1426, the measured Lag-time of Peak is compared against Target Lag Limits 1424. The measured lag-time should fall within the predetermined range (Target Lag Limits 1424) for the seal test to be valid. If the Lag-time of Peak is within the appropriate range then a logic 1 is provided to AND function 1432, otherwise a logic 0 is provided. In a second

comparison step 1428, the Magnitude of Peak is compared against a Peak\_threshold 1430. If the Magnitude of Peak is greater than the Peak\_threshold 1430 a logic 1 is provided to AND function 1432. This indicates that a significant portion of the acoustic test signal is present in the second volume measurement, otherwise a logic 0 is provided. A FAIL output 1434 corresponds to a logic 1 at the output of AND function 1432. The FAIL occurs when the Lag-time of Peak is within the predetermined range and the Magnitude of Peak is greater than the Peak\_threshold indicating that the acoustic barrier is sealed improperly. All other conditions indicate a PASS output 1434 and the acoustic barrier is sealed correctly.

*Id.*, 15:58-17:4.

**B. Prosecution History of the '493 Patent**

48. The '493 patent issued from U.S. Application 17/182,570, filed on 2/23/2021. The '493 patent is a continuation of U.S. Application 16/838,277 (now U.S. Patent No. 10,979,836), which in turn is a continuation of U.S. Application 16/414,136 (now U.S. Patent No. 10,701,499), which in turn is a continuation of U.S. Application 15/700,511 (now U.S. Patent No. 10,299,053), which in turn is a continuation of U.S. Application 14/827,332 (now U.S. Patent No. 9,781,530), which in turn is a continuation of U.S. Application 14/054,015 (now U.S. Patent No. 9,113,267), which in turn is a divisional of U.S. Application 12/555,864 (now U.S. Patent No. 8,600,067).

49. The '493 patent claims priority to U.S. Provisional Application No. 61/098,250, filed on 9/19/2008. Ex. 1005. I have been informed that the '493 patent is subject to the AIA because its parent, U.S. Patent No. 10,979,836, was prosecuted as an AIA application.

50. I have reviewed the prosecution history of the '493 patent and make the following observations.

51. The application that eventually issued as the '493 patent originally included 19 claims. EX1004, at 50-52. On February 14, 2022, applicants filed a preliminary amendment that amended originally-filed claims 1-19 and filed new claim 20. *Id.*, at 107-112. These amended claims are substantively the same as the issued claims.

52. The Office did not issue any rejections during prosecution. On November 16, 2022, the Office mailed a notice of allowance. *Id.*, 129 (Notice of Allowance). The Notice included an Examiner's amendment, which amended claim 1 as follows: "An audio device configured to test [[the]] a seal of an earphone..."; and "generating a test signal, wherein the test signal includes multiple frequency components, wherein at least one component lies between [[the]] a range of 30Hz and 200Hz." *Id.*, at 134. The Examiner also amended claim 16 to add a period to the end of the claim. *Id.*, 135.

53. In the Notice of Allowance issued on 12/23/2022, the Examiner stated the following “prior art made of record and not relied upon is considered pertinent to applicant’s disclosure”: “Ryan (US Pub. 20040196992),” “Siltmann (US Pub. 20070019817),” “Padhi (US Pub. 20050096899),” “Oliveira (US Pat. 5002151),” and “Rast (US Pub. 20010046304).” *Id.*, 222, 228-29. The Examiner stated that *Padhi* “discloses an apparatus, method, and computer program for comparing audio signals.” *Id.*, 229. The Examiner also considered the request for *ex parte* reexamination of the parent ’836 patent on 12/13/2022. *Id.*, 238. While the Examiner initialed *Voix* in a submitted IDS, he did not comment on *Voix*. *Id.* ’493FH, 376.

54. The Examiner’s search notes accompanying the notice of allowance listing submissions in an *inter partes* review (IPR) proceeding (IPR2022-00410) involving U.S. Patent No. 10,979,836 (the “’836 patent”), the immediate parent of the ’493 patent. *Id.*, 144. I note that I was also an expert witness retained by Samsung in that proceeding and submitted an expert declaration.

55. On November 18, 2022, Applicants requested continued examination for the Office to review Samsung’s *ex parte* reexamination request for the ’836 patent. *Id.*, 183-189, 210. I note that I was also an expert witness retained by Samsung in that proceeding and submitted an expert declaration.

56. The Office mailed a second Notice of Allowance on December 23, 2022. *Id.*, 222. In this Notice, the Examiner again identified *Svean* as the closest prior art to claim 1, and made identical statements regarding *Svean*. *Id.*, 227-28. The Examiner also stated that the following references were “considered pertinent to applicant’s disclosure” and were “made of record and not relied upon”: Ryan (U.S. Pub. 20040196992), Siltmann (U.S. Pub. 20070019817), Padhi (U.S. Pub. 20050096899), Oliveira (US Pat. 5002151), and Rast (US Pub. 20010046304). *Id.*, 228-29.

57. Regarding *Padhi*, the Examiner stated that it “discloses an apparatus, method, and computer program for comparing audio signals.” *Id.*, 229.

58. The ’493 patent issued on May 30, 2023. *Id.*, 380-387.

### **C. The Claims**

59. The table below includes identifiers for the ’493 patent claims and limitations, which are referenced throughout.

| <b>Claim 1</b> |                                                                                                                                |
|----------------|--------------------------------------------------------------------------------------------------------------------------------|
| 1.pre          | An audio device configured to test a seal of an earphone comprising:                                                           |
| 1.1            | a memory that stores instructions; and                                                                                         |
| 1.2            | a processor, wherein the processor is configured to execute the instructions to perform operations, the operations comprising: |
| 1.2.1          | generating a test signal,                                                                                                      |

|                |                                                                                                                                                                                                                                                                                                                                  |
|----------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| 1.2.1.1        | wherein the test signal includes multiple frequency components, wherein at least one component lies between a range of 30 Hz and 200 Hz;                                                                                                                                                                                         |
| 1.2.2          | sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user, |
| 1.2.3          | receiving a first microphone signal from the first microphone;                                                                                                                                                                                                                                                                   |
| 1.2.4          | calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination;                                                                                                    |
| 1.2.5          | comparing the value to a threshold to determine a seal quality between the first side and the second side;                                                                                                                                                                                                                       |
| 1.2.6          | generating a message to convey the seal quality; and                                                                                                                                                                                                                                                                             |
| 1.2.7          | sending the message to the user.                                                                                                                                                                                                                                                                                                 |
| <b>Claim 2</b> |                                                                                                                                                                                                                                                                                                                                  |
| 2              | The audio device according to claim 1, wherein the message is an acoustic message.                                                                                                                                                                                                                                               |
| <b>Claim 3</b> |                                                                                                                                                                                                                                                                                                                                  |
| 3              | The audio device according to claim 1, wherein the message is a visual message sent to a display of the audio device.                                                                                                                                                                                                            |
| <b>Claim 4</b> |                                                                                                                                                                                                                                                                                                                                  |
| 4              | The audio device according to claim 1, wherein the speaker plays the test signal from the first side.                                                                                                                                                                                                                            |
| <b>Claim 5</b> |                                                                                                                                                                                                                                                                                                                                  |
| 5              | The audio device according to claim 4, wherein the first microphone measures sound from the first side.                                                                                                                                                                                                                          |

| <b>Claim 6</b>  |                                                                                                                                                                                                                                    |
|-----------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| 6               | The audio device according to claim 4, wherein the first microphone measures sound from the second side.                                                                                                                           |
| <b>Claim 7</b>  |                                                                                                                                                                                                                                    |
| 7               | The audio device according to claim 1, wherein the first signal is the test signal and the second signal is the first microphone signal.                                                                                           |
| <b>Claim 8</b>  |                                                                                                                                                                                                                                    |
| 8               | The audio device of claim 1, wherein the first signal is the test signal and the second signal is a modified first microphone signal.                                                                                              |
| <b>Claim 9</b>  |                                                                                                                                                                                                                                    |
| 9               | The audio device according to claim 8, wherein the modified first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200 Hz. |
| <b>Claim 10</b> |                                                                                                                                                                                                                                    |
| 10              | The audio device of claim 1, wherein the first signal is a modified test signal and the second signal is the first microphone signal.                                                                                              |
| <b>Claim 11</b> |                                                                                                                                                                                                                                    |
| 11              | The audio device according to claim 1, wherein the first signal is a modified test signal and the second signal is a modified first microphone signal.                                                                             |
| <b>Claim 12</b> |                                                                                                                                                                                                                                    |
| 12              | The audio device according to claim 11, wherein the first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200 Hz.         |
| <b>Claim 13</b> |                                                                                                                                                                                                                                    |
| 13              | The audio device according to claim 1, wherein the earphone includes a second microphone.                                                                                                                                          |
| <b>Claim 14</b> |                                                                                                                                                                                                                                    |

|                 |                                                                                                                                                            |
|-----------------|------------------------------------------------------------------------------------------------------------------------------------------------------------|
| 14              | The audio device according to claim 13, wherein the operations further comprise: receiving a second microphone signal from the second microphone.          |
| <b>Claim 15</b> |                                                                                                                                                            |
| 15              | The audio device according to claim 14, wherein the first signal is the first microphone signal and the second signal is the second microphone signal.     |
| <b>Claim 16</b> |                                                                                                                                                            |
| 16              | The audio device of claim 14, wherein the first signal is a modified first microphone signal and the second signal is the second microphone signal.        |
| <b>Claim 17</b> |                                                                                                                                                            |
| 17              | The audio device of claim 14, wherein the first signal is a modified first microphone signal and the second signal is a modified second microphone signal. |
| <b>Claim 18</b> |                                                                                                                                                            |
| 18              | The audio device of claim 1, wherein the seal quality is good if the value is at or above the threshold value.                                             |
| <b>Claim 19</b> |                                                                                                                                                            |
| 19              | The audio device of claim 1, wherein the sealing section is composed of at least one of rubber, silicone, a flexible polymer, or a combination thereof.    |
| <b>Claim 20</b> |                                                                                                                                                            |
| 20              | The audio device of claim 1, wherein the sealing section is an eartip.                                                                                     |

## VIII. CLAIM CONSTRUCTION

60. I understand that as to the parent '836 patent of the '493 patent and an unrelated patent Patent Owner has asserted against Samsung (U.S. Patent

9,609,424), the Eastern District of Texas construed the following terms, which also appear in the challenged claims of the '493 patent:

- “***a processor . . . the processor***” ('836 patent, claims 1, 9, 14): “The same processor must perform each of the recited functions.”

EX1017, 48.

- “***seal quality***” ('424 patent, claim 1): “[T]he Court agrees with Samsung[’s] position that ‘seal quality’ is not binary in nature. Otherwise, this term will be given its plain and ordinary meaning.”

EX1017, 39, 50.

61. I further understand that, in the currently pending *ex parte* reexamination of the '836 patent (90/015,146), the Examiner asserted in a final rejection that the term “the processor is configured to execute the instructions to perform operations, the operations comprising” should be interpreted subject to §112(f), and therefore is a means-plus-function limitation. EX1014, 8-9. Patent Owner has appealed and disputed that the term is subject to §112(f). EX1018. To the extent that the term “processor” is subject to §112(f), and to the extent there is corresponding structure disclosed in the '493, that structure includes a processor or application specific integrated chip (ASIC) ('493, 5:53-67, (“processor 206 can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or digital signal processor (DSP) with associated

Random Access Memory (RAM) 202 and Read Only Memory (ROM) 204”)) performing the algorithm illustrated in Figs. 3 and 4, which includes generating a test signal (step 304), emitting the test signal to a speaker (step 306), obtaining a microphone signal from a microphone (step 308), calculating a value using cross-correlation (step 310/412), comparing the value to a threshold (step 414), and generating a message and sending the message to the user (step 418/420). ’493, 8:6-40, 9:15-52. A similar algorithm applicable to cross-correlation, correlation and coherence is described in the context of Fig. 14. ’493, 16:6-17:4. As explained in detail below, the prior art discloses a processor implementing this algorithm, using cross-correlation.

62. Here, I have given all the claim terms of the challenged claims their plain and ordinary meaning, as would have been understood by a person having ordinary skill in the art at the time of the alleged invention, having taken into consideration the language of the claims, the specification, and the prosecution history of record. At the request of counsel, I have also considered in the alternative “the processor is configured to execute the instructions to perform operations, the operations comprising” limitation as a means-plus-function limitation.

63. For purposes of this declaration, I will assume that the time of the alleged invention is 9/19/2008 (the filing date of U.S. Provisional Patent Application 61/098,250).

## IX. SPECIFIC GROUNDS FOR CHALLENGE

### A. Ground 1: *Epley, Padhi and Voix* Render Obvious Claims 1, 3-5, 7-8, 10-11, 18-20

64. I have reviewed *Epley, Padhi, and Voix*, and as described below, it is my opinion that *Epley* in combination with *Padhi* and *Voix* renders obvious all of the features of claims 1, 3-5, 7-8, 10-11, 18-20.

#### 1. Overview of *Epley*

65. *Epley* is directed to a “Stimulus-Evoked Vestibular Evaluation (SEVE)” system for detection of vestibular (inner ear) disorders. *Epley*, [0015].

The invented ***Stimulus-Evoked Vestibular Evaluation (SEVE)*** or Tullimeter™ System is a system for ***detection and analysis of stimulus-evoked (by sound or pressure) manifestations of vestibular disorders***, to be utilized in the evaluation of subjects with a complaint of dizziness or imbalance. It is designed as a practical, but comprehensive, diagnostic tool for provocative testing of sound and pressure evoked vestibular phenomena (postural destabilization and abnormal ocular phenomena) found in certain inner ear conditions that cause vertigo and imbalance. Tullimeter™ is a trademark worldwide rights in which are claimed by and reserved to Epley Research LLC.

*Epley*, [0015].

66. The SEVE system includes **ear probes** and a “sound/pressure generator/feedback mechanism 24.” *Epley*, [0066].

A described above, headphones can be used instead of **ear probes 18** to deliver sound (but not pressure) to the subject's ears. Those of skill will appreciate that such a configuration of system 10 would require only parts (i.e. those including the sound generator circuitry of **sound/pressure generator/feedback mechanism 24** to be active during subject testing. Such an alternative configuration of ear stimulus equipment nevertheless permits sound stimulation of a subject and ocular and/or sway response that might indicate vestibular function anomaly or pathology. This and other alternatives to the use of insertable and sealingly engageable ear probes 18 for sound and/or pressure testing all are contemplated as being within the spirit and scope of the invention.

*Epley*, [0066].

67. The **ear probes** are “inserted into a subject’s ear,” **seal** an ear, and play audio as part of diagnosis. *Epley*, [0004].

The invented apparatus for testing a subject's vestibular response to ear pressure or sound stimuli includes a headset including one or more ear-worn devices such a headphones or probes, wherein **each probe is configured to be inserted into a subject's ear in an operational alignment with the subject's ear canal and sealing engagement with the wall of the subject's ear canal, and a stimulus-producing mechanism operatively coupled with the headset configured to**

***deliver defined pressure or sound stimuli to the one or more probes.***

The apparatus further includes one or both of 1) an ocular response analyzer operatively coupled to the headset, the ocular response analyzer measuring the subject's nystagmatic response to the stimuli by electronic, e.g. videographic, means and computerized comparison of potentially aberrational response data from the subject to recorded baseline subject data and 2) a sway response analyzer operatively coupled to the headset, the sway response analyzer including an inertial sensor of the subject's head and torso positional and orientational response to the stimuli and computerized comparison of potentially aberrational inertial data from the subject to recorded baseline inertial subject data. An optional destabilizing device for producing imbalance in a nominally stable subject thereby to test the subject's vestibular function and a carrying case configured and dimensioned portably to contain and transport the headset including the probes, the stimulus-producing mechanism, the ocular response analyzer and the destabilizing device therein.

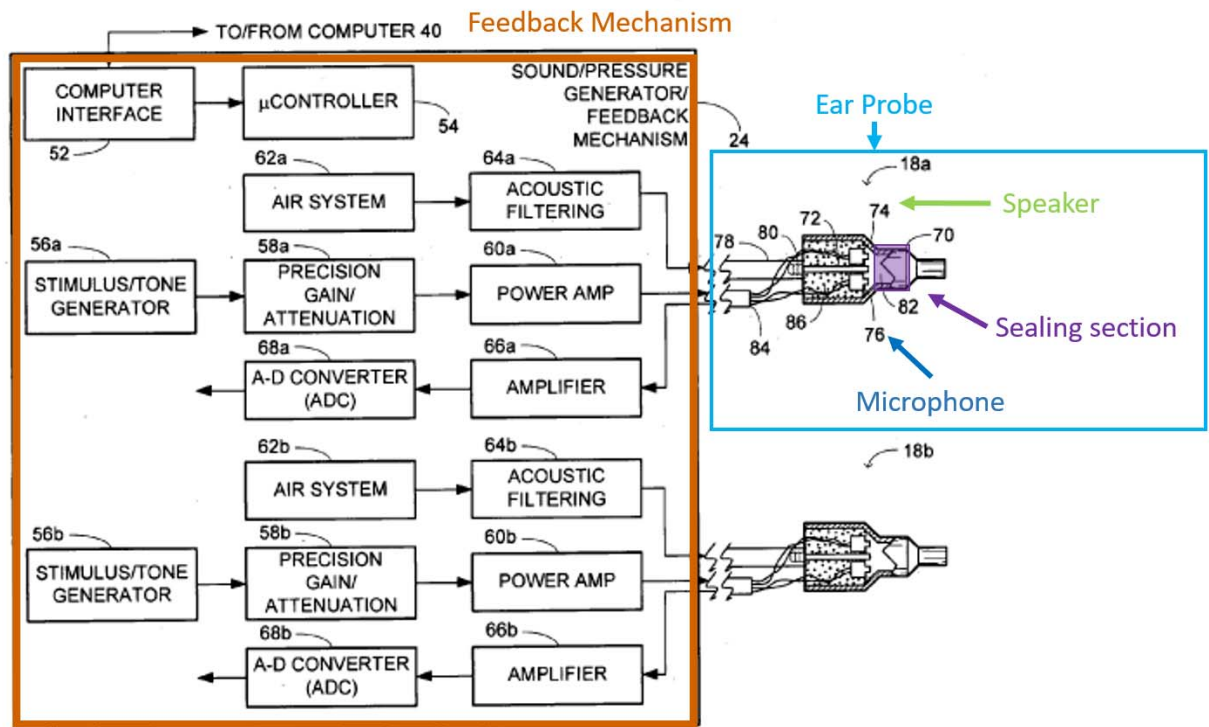
*Epley, [0004].*

68. Each ear probe includes a “speaker 74 and microphone 76.” *Epley, [0065], Figs. 2, 3.*

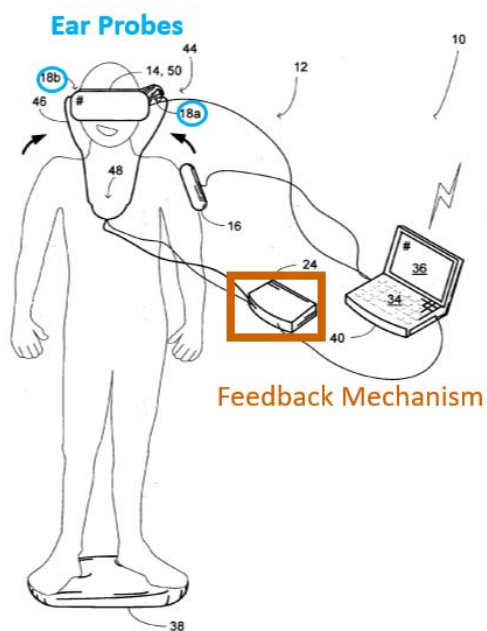
In accordance with one embodiment of the invention, air tube 72, ***speaker 74 and microphone 76 are secured within earpiece 70 by potting compound 86, e.g. of epoxy resin.*** In accordance with one embodiment of the invention, potting compound 86 extends only

partway toward the distal end of earpiece 70, thus forming hollow (air-filled) distal region 82 that is in fluid (air) communication with the subject's left ear. Alternatively, yet within the spirit and scope of the invention, the potting compound can be of any suitable material that instead may fill the entire interior region of earpiece 70. In accordance with one embodiment of the invention, earpiece 70 is of machined or extruded aluminum, although within the spirit and scope of the invention it can be of any suitable material. Those of skill in the art will appreciate that earpiece 70 is configured to have an ear-canal-easy cushion, not shown, slipped over its distal end for the comfort of a test subject.

*Epley*, [0065]



Epley, Figure 3 (annotated)



Epley, Figure 2 (annotated)

69. Epley teaches, “to ensure proper sealing of **left** and **right ear probes 18a and 18b**,” the “**feedback mechanism**” tests the seal “by applying a known sound and volume and measuring the acoustic response within the subjects ear as picked up by the microphone,” and a “straightforward comparison of input and output sound parameters determine whether the **ear probe** is properly sealed.” Epley, [0070]. A “display produces a warning to the operator” when appropriate. Epley, [0070].

Those of skill in the art will appreciate that *microphone 76, amplifier 66 a/b and ADC 68 a/b that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ensure proper sealing of **left and right ear probes. 18 a and 18 b**. This is accomplished, for example, by applying a known sound and volume and measuring the acoustic response within the subject's ear as picked up by the microphone. A straightforward comparison of input and output sound parameters determine whether the **ear probe** is properly sealed and further testing is avoided if the seal is not secure. The display produces a warning to the operator.* Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

Epley, [0070]

## 2. Overview of Padhi

70. Padhi is directed to a system for “comparing audio signals.” *Padhi*, [0001].

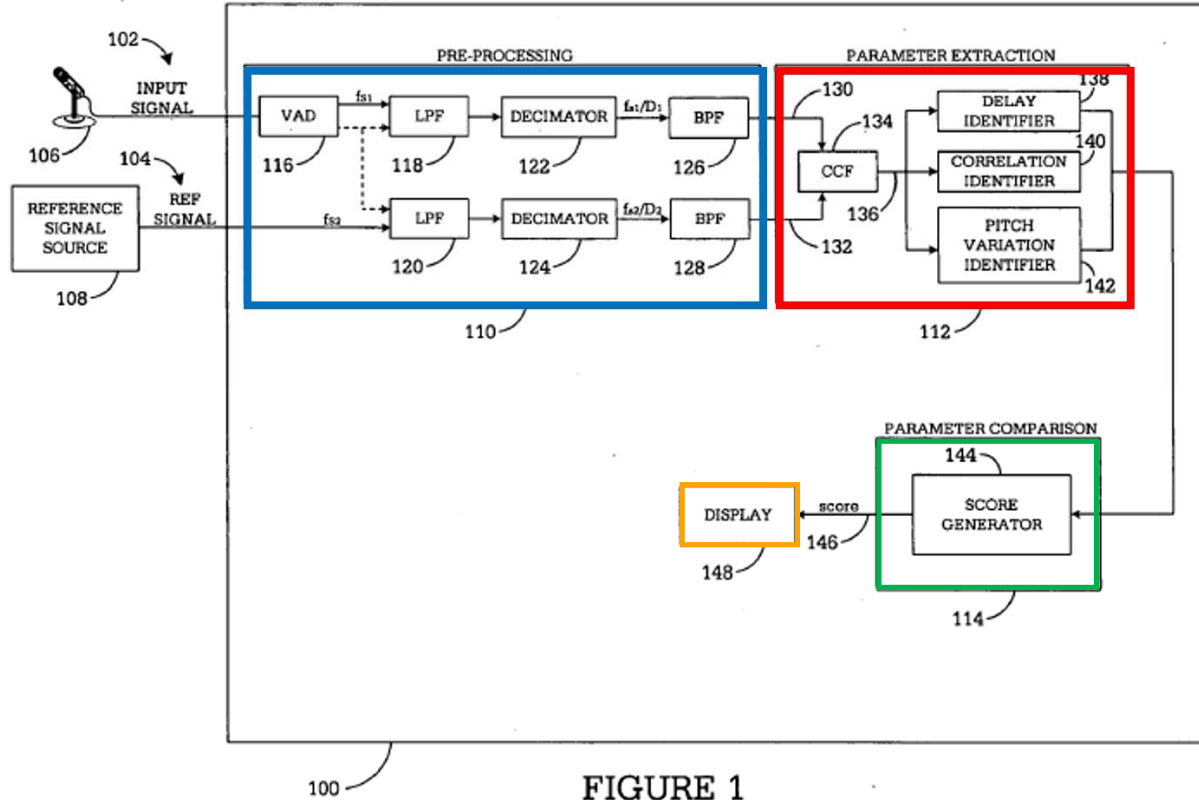
This disclosure is generally directed to audio systems and more specifically to an ***apparatus, method, and computer program for comparing audio signals.***%

*Padhi*, [0001]. *Padhi* discloses “receiving and cross-correlating a first audio signal and a second audio signal.” *Padhi*, Abstract.

An apparatus, method, and computer program are capable of ***receiving and cross-correlating a first audio signal and a second audio signal.*** This produces a cross-correlated signal, which is used to identify a plurality of parameters associated with at least one of the first and second audio signals. The parameters are used to generate an indicator identifying an extent to which the first and second audio signals match.

*Padhi*, Abstract.

71. Figure 1 illustrates *Padhi*’s apparatus, which includes **pre-processing section 110**, **parameter extraction section 112**, **parameter comparison section 114**, and **display 148**. *Id.*, [0020]; *id.*, [0033].



72. *Padhi* discloses that “various components shown in FIG. 1 may represent ... software routines executed by one or more processors.” *Padhi*, [0034].

*The various components shown in FIG. 1 may represent any suitable hardware, software, firmware, or combination thereof.* As an example, the components in the pre-processing section 110, parameter extraction section 112, and parameter comparison section 114 may represent *software routines executed by one or more processors* in the apparatus 100.

*Padhi*, [0034].

73. *Padhi* goes on to state that “various changes may be made to Fig. 1,” and “[v]arious components can be omitted or combined and additional components can be added according to particular needs.” *Padhi*, [0035].

Although FIG. 1 illustrates one example of an audio apparatus 100, various changes may be made to FIG. 1. For example, the functional division of the apparatus 100 is for illustration only. ***Various components can be omitted or combined and additional components can be added according to particular needs.*** As particular examples, ***the voice activity detector 116 could be omitted***, and the reference signal source 108 could form part of the apparatus 100. Also, the filters 118, 126 may be combined, and the filters 120, 128 may be combined. Further, the input signal 102 and reference signal 104 could represent analog or digital signals. If one or both signals 102, 104 are analog signals, one or more amplifiers and/or one or more analog-to-digital converters could be used to convert the signals 102, 104 into a digital format. In addition, the apparatus 100 could form part of a larger apparatus or system, such as a karaoke system or a computer system.

*Padhi*, [0035].

74. In Figure 1, the apparatus 100 receives an input audio signal 102 via a microphone 106. A reference signal 104 is supplied by a reference signal source 108. *Padhi* explains that, however, these signals “could be replaced by any suitable audio signals provided by any suitable source or sources.” *Padhi*, [0018].

In the illustrated example, ***the apparatus 100 receives an input signal 102 and a reference signal 104.*** The input signal 102 and the reference signal 104 represent audio signals to be compared by the apparatus 100. ***In this example, the input signal 102 is supplied by a microphone 106, and the reference signal 104 is supplied by a reference signal source 108.*** In a particular embodiment, the input signal 102 is produced by a microphone 106 used by a singer, and the reference signal 104 represents an original song provided by a compact disc (CD) player. In another particular embodiment, the input signal 102 is produced by a microphone 106 used by a student, and the reference signal 104 represents a proper pronunciation of words to be spoken by the student. This represents two possible uses of the apparatus 100. ***However, the illustrated input signal 102 and reference signal 104 could be replaced by any suitable audio signals provided by any suitable source or sources.***

*Padhi*, [0018].

75. Prior to performing cross correlation, the microphone and reference signals may be pre-processed via low pass filtering, decimating, and/or band pass filtering the signals. For example:

In the pre-processing section 110, the input signal 102 is supplied to a ***filter 118***, and the reference signal 104 is supplied to a filter 120. The filters 118, 120 filter the signals 102, 104 before the signals 102, 104 are processed further. Each filter 118, 120 includes any hardware, software, firmware, or

combination thereof for **filtering a signal**. As an example, **the filters 118, 120 may represent anti-aliasing low pass filters**. As a particular example, **the filters 118, 120 may represent anti-aliasing low pass filters having a cut-off frequency of 3 kHz**. In some embodiments, the filters 118, 120 do not process the signals 102, 104 until the voice activity detector 116 detects a voice in the input signal 102.

*Padhi*, [0023];

**A filter 126 is coupled to the decimator 122, and a filter 128 is coupled to the decimator 124. The filters 126, 128 filter the decimated input signal 102 and decimated reference signal 104.** The filters 126, 128 filter the decimated signals 102, 104 before the decimated signals 102, 104 are processed further. **Each filter 126, 128 includes any hardware, software, firmware, or combination thereof for filtering a signal**. As an example, the filters 126, 128 may represent **band pass filters**. As a particular example, the filters 126, 128 may represent infinite impulse response (IIR) filters that band limit the decimated signals 102, 104 to 1 kHz, which helps to attenuate frequencies above 1.5 kHz.

*Padhi*, [0025].

76. The apparatus cross-correlates the two signals to determine the extent to which they match. *Padhi.*, [0027].

The parameter extraction section 112 includes a **cross correlator** (CCF) 134. The cross correlator 134 receives the pre-processed

signals 130, 132 and cross-correlates the pre-processed signals 130, 132. This produces a cross-correlated signal 136. The cross correlator 134 may use any suitable technique to cross correlate the pre-processed signals 130, 132. The cross correlator 134 includes any hardware, software, firmware, or combination thereof for cross-correlating two or more signals.

*Padhi.*, [0027]. *Padhi* describes that the “cross correlation may be used to identify an amount of correlation between the signals” and “indicator identifying the extent to which the signals, 102, 104 match.” *Padhi.*, [0017].

In one aspect of operation, the apparatus 100 receives audio signals, such as an input signal 102 and a reference signal 104. The apparatus 100 cross-correlates the audio signals and uses the cross correlation to identify various characteristics or parameters of the audio signals. For example, the cross correlation may be used to identify a variation of pitch between the signals 102, 104. The cross correlation may also be used to identify a delay between the signals 102, 104. ***In addition, the cross correlation may be used to identify an amount of correlation between the signals 102, 104. Using the identified characteristics, the apparatus 100 generates a score or other indicator identifying the extent to which the signals 102, 104 match.***

*Padhi.*, [0017]. The cross-correlation is used to identify various characteristics associated with the two signals. *Padhi.*, [0028].

The cross-correlated signal 136 is provided to a delay identifier 138, a correlation identifier 140, and a pitch variation identifier 142. The identifiers 138-142 use the cross-correlated signal 136 to identify various characteristics associated with the input signal 102 and/or the reference signal 104.

*Padhi.*, [0028]. *Padhi* further discloses that the cross-correlation identifies the amount of correlation between the signals 102, 104. *Padhi.*, [0030].

The correlation identifier 140 ***identifies the amount of correlation between the signals 102, 104*** using the cross-correlated signal 136. For example, the reference signal 104 could represent a song, and the amount of correlation between the signals 102, 104 represents whether a singer sung the correct lyrics in the song. In some embodiments, the correlation identifier 140 identifies the amount of correlation between the signals 102, 104 in the form of a percentage. The correlation identifier 140 uses any suitable technique for identifying the amount of correlation between signals. The correlation identifier 140 includes any hardware, software, firmware, or combination thereof for identifying the amount of correlation between two or more signals. As a particular example, the correlation identifier 140 may include a normalizer that normalizes the cross-correlated signal 136. ***The normalized value of the cross-correlated signal 136 represents an indication of the percentage match between the signals 102, 104.***%

*Padhi*, [0030]. *Padhi*'s system identifies the extent to which the signals match.

*Padhi*, [0071].

The apparatus 100 uses the identified parameters to generate a score at step 610. ***The score identifies the extent to which the signals 102, 104 match.*** This may include, for example, the score generator 144 receiving the identified parameters from the identifiers 138-142 and generating the score using the parameters.

*Padhi*, [0071].

77. The amount of correlation between the signals is a component of an overall “score.” *Padhi*, [0032]

The identified delay, amount of correlation, and pitch variation are supplied to the parameter comparison section 114. The parameter comparison section 114 includes a score generator 144. The score generator 144 processes the identified delay, correlation, and pitch variation and generates a score 146 using these values. ***The score 146 represents an indicator identifying an extent to which the signals 102, 104 match.*** The score 146 could, for example, represent a value between 0 and 100. In some embodiments, the input signal 102 and the reference signal 104 may be divided into frames, and a score 146 may be produced for each frame. The score generator 144 may also dynamically adjust the score 146 for one frame based on previous frames. The score generator 144 uses any suitable technique for generating a score 146. One technique for generating a score 146 is described below with respect to FIG. 6. The score generator 144

includes any hardware, software, firmware, or combination thereof for generating a score 146 using the information generated by the parameter extraction section 112. Although this has described the score generator 144 using the identified delay, correlation, and pitch variation to generate a score 146, other or additional characteristics could also be used to generate a score 146.

*Padhi*, [0032]. The apparatus presents the score to an end user via a **display**.

*Padhi*, [0033].

***The score 146 produced by the score generator 144 is presented to a user or other person or people on a display 148.***

***The display 148 represents any suitable device or structure for presenting a score 146.*** Although FIG. 1 illustrates the display 148 as forming part of the apparatus 100, the display 148 could reside external to the apparatus 100. Also, the score 146 could be used in other ways, such as by storing the score 146 in a database or other storage medium.

*Padhi*, [0033]. **Display** 148 represents “any suitable device or structure for presenting a score.” *Padhi*, [0033].

78. *Padhi* discloses more details regarding an example technique for generating a score with reference to Figure 6.

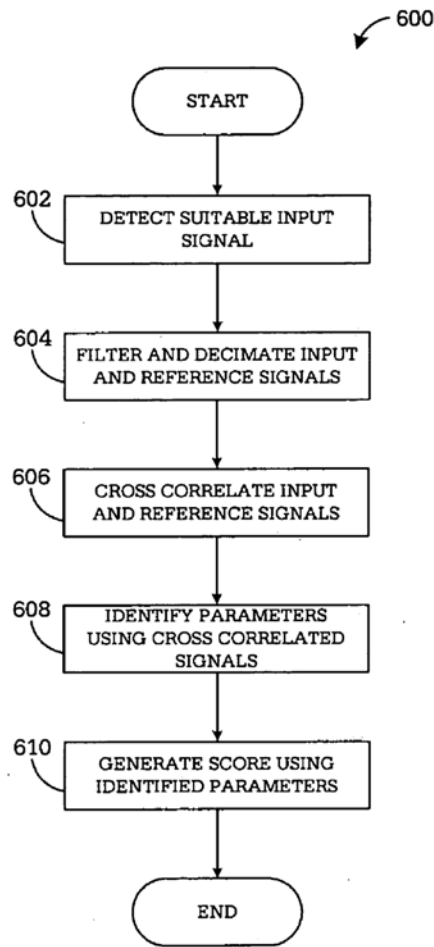


FIGURE 6

Padhi, Fig. 6.

79. At step 606, the signals are cross-correlated. *Padhi*, [0069].

***The apparatus 100 cross-correlates the processed signals at step 606.***

This may include, for example, the cross correlator 134 correlating the pre-processed input and reference signals 130, 132 to produce a cross-correlated signal 136. The cross correlator **134** may correlate an equal

number of samples from the pre-processed signals 130, 132. The cross correlator 134 could also correlate an unequal number of samples from the pre-processed signals 130, 132.

*Padhi* [0069]. At step 610, the score is generated to identify the extent to which the signals match. *Padhi* [0071].

The apparatus 100 uses the identified parameters to generate a score at step 610. ***The score identifies the extent to which the signals 102, 104 match.*** This may include, for example, the score generator 144 receiving the identified parameters from the identifiers 138-142 and generating the score using the parameters.

*Padhi* [0071]. *Padhi* describes that “various changes may be made to Fig. 6,” for example, as I noted above, more or less pre-processing.

Although FIG. 6 illustrates one example of a method 600 for comparing audio signals, ***various changes may be made to FIG. 6.*** For example, the apparatus 100 could always process audio signals and not wait to detect a suitable input signal 102 at step 602.%

*Padhi* [0079].

### 3. Motivation to apply *Padhi*'s teachings to *Epley*

80. In my opinion, *Padhi* and *Epley* are analogous prior art. It is my opinion that *Padhi* and *Epley* are in the same field as the '493 patent, which includes audio systems. Moreover, *Padhi* and *Epley* are reasonably pertinent to alleged problems addressed by the '493 patent of comparing signals for audio applications. For example, the '493 patent explains that its embodiments are directed to "acoustic devices." '493 patent, 3:32-37

Exemplary embodiments are directed to or can be operatively used on various wired or wireless orifice inserted devices for example earpiece devices (e.g., earbuds, headphones, ear terminals, behind the ear devices or other acoustic devices as known by one of ordinary skill, and equivalents).

'493 patent, 3:32-37. Likewise, *Epley* describes its invention as used to "invention to ensure proper sealing of" earpieces. *Epley*, [0070].

Those of skill in the art will appreciate that microphone 76, amplifier 66 a/b and ADC 68 a/b that form part of the feedback mechanism are used in accordance with one embodiment of the invention **to ensure proper sealing of left and right ear probes. 18 a and 18 b**. This is accomplished, for example, by applying a known sound and volume and measuring the acoustic response within the subject's ear as picked up by the microphone. **A straightforward comparison of input and output sound parameters determine whether the ear probe is properly sealed and further testing is avoided if the seal is not**

**secure.** The display produces a warning to the operator. Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

*Epley*, [0070]. And like *Epley*, *Padhi* describes a comparison of acoustic signals to determine the extent to which the signals match. *Padhi*, [0032].

The identified delay, amount of correlation, and pitch variation are supplied to the parameter comparison section 114. The parameter comparison section 114 includes a score generator 144. The score generator 144 processes the identified delay, correlation, and pitch variation and generates a score 146 using these values. ***The score 146 represents an indicator identifying an extent to which the signals 102, 104 match.*** The score 146 could, for example, represent a value between 0 and 100. In some embodiments, the input signal 102 and the reference signal 104 may be divided into frames, and a score 146 may be produced for each frame. The score generator 144 may also dynamically adjust the score 146 for one frame based on previous frames. The score generator 144 uses any suitable technique for generating a score 146. One technique for generating a score 146 is described below with respect to FIG. 6. The score generator 144 includes any hardware, software, firmware, or combination thereof for generating a score 146 using the information generated by the parameter extraction section 112. ***Although this has described the score generator 144 using the identified delay, correlation, and pitch***

*variation to generate a score 146, other or additional characteristics could also be used to generate a score 146.*

*Padhi*, [0032]. Figure 1 of *Padhi* further illustrates *Padhi*'s signal computations.

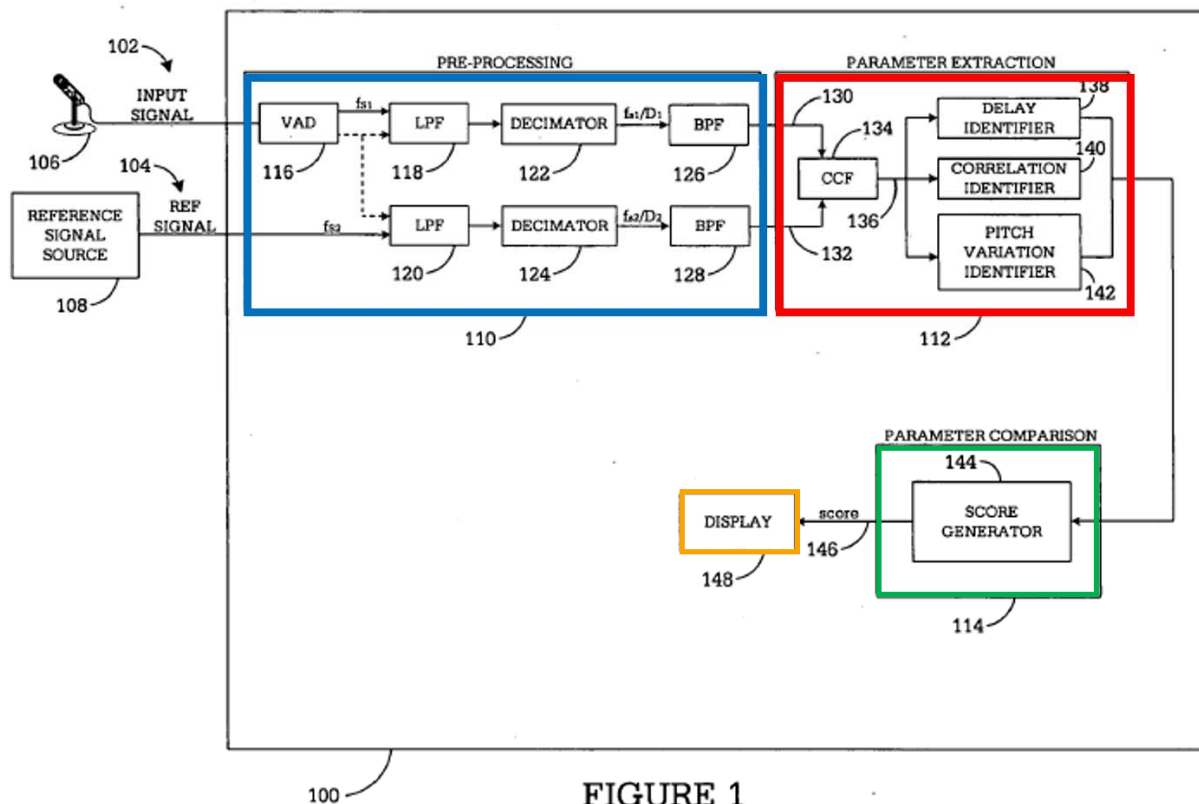


FIGURE 1

Consistent with this, in the file history of the '493 patent, the Examiner recognized *Padhi* as "pertinent art." '493FH at 228. *Padhi* is directed to techniques for "comparing" the similarity of two "audio signals." *Padhi*, [0001] ("This disclosure is generally directed to audio systems and more specifically to an apparatus, method, and computer program for comparing audio signals."), [0004] ("This disclosure provides an apparatus, method, and computer program for

comparing audio signals.”), (“In one aspect, an apparatus includes a cross correlator operable to receive a first audio signal and a second audio signal and to cross correlate the first and second audio signals, producing a cross-correlated signal.”), [0006] (“In another aspect, a method includes receiving a first audio signal and a second audio signal and cross-correlating the first and second audio signals to produce a cross-correlated signal.”). This is pertinent to the acoustic seal analysis problem addressed by the ’493 patent because—as references such as *Epley* disclose—assessing acoustic seal involves a determination of the similarity of audio signals. *Epley*, [0070] (“[C]omparison of input and output sound parameters determine whether the ear probe is properly sealed.”)

81. In my opinion, a person of ordinary skill in the art would have been motivated to apply *Padhi*’s comparison of a reference signal and microphone signal through “cross-correlation” to obtain a value  $\alpha$  to *Epley*’s audio device for several reasons.

82. First, *Epley* leaves the details of its “comparison of input and output sound parameters” for its seal test to the person of ordinary skill in the art to implement, and thus a person of ordinary skill in the art would have been motivated to look to analogous art—including *Padhi*—for how to carry out such comparison of sound parameters. For example, *Epley* simply refers to the

“comparison of input and output sound parameters,” but does not provide details.

*Epley*, [0070].

Those of skill in the art will appreciate that microphone 76, amplifier 66 a/b and ADC 68 a/b that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ensure proper sealing of left and right ear probes. 18 a and 18 b. This is accomplished, for example, by applying a known sound and volume and measuring the acoustic response within the subject's ear as picked up by the microphone. ***A straightforward comparison of input and output sound parameters determine whether the ear probe is properly sealed and further testing is avoided if the seal is not secure.*** The display produces a warning to the operator. Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

*Epley*, [0070]. And *Padhi* explains comparing a microphone signal and a known reference signal through a cross-correlation calculation. *Padhi*, [0027] (“The parameter extraction section 112 includes a cross correlator (CCF) 134. The cross correlator 134 receives the pre-processed signals 130, 132 and cross-correlates the pre-processed signals 130, 132. This produces a cross-correlated signal 136”).

83. Consistent with this, the Examiner in the *EPR* of parent '836 patent found that *Padhi*'s cross correlation was applicable to a seal test and that a cross correlation “was a notoriously well-known method of comparing audio signals at

the time of the invention.” EX1014, 28. The same reasoning applies here, because *Padhi*’s cross correlation compares two signals by calculating a value with the two signals.

84. Second, *Padhi* explains that its cross-correlation technique is an improvement over other audio signal comparisons, “which do[] not provide an accurate comparison of the actual contents of the signals” and “are often computationally intensive.” *Padhi*, [0003].

Some of these audio systems use simplistic algorithms for comparing audio signals. For example, some audio systems simply compare the frame energies of the audio signals. In effect, ***these systems compare the loudness of the audio signals, which does not provide an accurate comparison of the actual contents of the signals. Other audio systems use overly complex algorithms, such as fast Fourier transformation techniques. These techniques are often computationally intensive and do not compare the actual contents of the audio signals.***

85. Third, *Padhi*’s comparison of a reference signal and input microphone signal is directly applicable to Epley’s comparison of a “known sound” (“output parameter”) and “the acoustic response...as picked up by the microphone” (“input parameter”). See, e.g., *Padhi*, [0027]-[0029] (explaining that the cross-correlation is performed on the “input signal 102 and the reference signal 104” illustrated in Figure 6); *Epley*, [0070] (“This is accomplished, for example, by applying a known

sound and volume and measuring the acoustic response within the subject's ear as picked up by the microphone. A straightforward comparison of input and output sound parameters determine whether the ear probe is properly sealed and further testing is avoided if the seal is not secure.”).

86. In light of the above, in my opinion, a person having ordinary skill in the art would have found it routine, straightforward and advantageous to apply **Padhi’s** teachings of cross-correlating a reference signal and an input signal to **Epley’s** teachings of a comparing a known sound with a microphone signal, and would have known that such a combination (yielding the claimed limitations) would predictably work and provide the expected functionality.

#### **4. Overview of Voix**

87. *Voix* teaches “an apparatus for assessing in situ an acoustic seal of an in-ear device.” *Voix*, [0092] (“FIG. 1 is a simplified schematic diagram of an apparatus for assessing in-situ an acoustic seal of an in-ear device in accordance with an embodiment of the present invention”). The apparatus includes “sound source 24 provid[ing] a sound typically having a frequency range varying from about 100 Hz to about 10000 Hz.” *Voix*, [0100].

The apparatus 10 typically includes a controller unit 22 operatively connected to a broadband reference sound source 24 to control operation thereof. ***The sound source 24 provides a sound typically having a frequency range varying from about 100 Hz to about***

**10000 Hz such that the following preferred octave bands which enclose most audible sounds** (including conversation frequency range of about 300 Hz to about 3000 Hz) **are essentially covered:** 125 Hz, 250 Hz, 500 Hz, 1000 Hz, 2000 Hz, 4000 Hz and 8000 Hz.

Voix, [0100].

88. To assess an acoustic seal, “a first sound level outside the ear canal” and “a second sound level inside the ear canal” are measured, and the second sound level is subtracted from the first sound level. Voix, [0036].

In accordance with an aspect of the present invention, there is provided a method for assessing in-situ an acoustic seal of an in-ear device using an apparatus having a sound measurement device connected thereto, the in-ear device having a passageway extending therethrough, the passageway being in fluid communication with an ear canal of an ear when the in-ear device is inserted therein, the sound measurement device being removably engageable into the passageway, said method comprises: a) calibrating said sound measurement device by measuring a reference sound level with said sound measurement device when being submitted to a reference sound source and when being located in a close relationship relative thereto; b) **measuring a first sound level outside the ear canal** with said sound measurement device when submitted to the sound source and when located in a close relationship relative to the in-ear device and outside the ear canal; c) **measuring a second sound level inside the ear canal** with said sound measurement device when submitted to the

sound source and when engaged into and occluding the passageway with the in-ear device inserted inside the ear canal; and d) ***assessing the acoustic seal of the in-ear device by subtracting said second sound level from said first sound level.***

89. Voix teaches “indicating presence of acoustic seal when attenuation between said first and second sound levels is equal to or larger than a predetermined threshold value within a predetermined frequency range.” Voix, [0039].

#### **5. Motivation to apply Voix’s teachings to *Epley* and *Padhi***

90. In my opinion, like *Epley* and *Padhi*, Voix is analogous prior art and in the same field of acoustic devices as the ’493 patent, and is reasonably pertinent to alleged problems addressed by the ’493 patent of comparing signals for audio applications. For example, the ’493 patent explains that its embodiments are directed to “acoustic devices.” ’493 patent, 3:32-37.

Exemplary embodiments are directed to or can be operatively used on various wired or wireless orifice inserted devices for example earpiece devices (e.g., earbuds, headphones, ear terminals, behind the ear devices or other acoustic devices as known by one of ordinary skill, and equivalents).

’493 patent, 3:32-37.

91. Likewise, *Voix* describes a method and apparatus for assessing the seal of an in-ear device. *Voix*, [0036] (“In accordance with an aspect of the present invention, there is provided a method for assessing in-situ an acoustic seal of an in-ear device...”), [0044] (“In one embodiment, step d) includes assessing the acoustic seal of the in-ear device by subtracting said second sound level from said first sound level and using a compensation term relating to the in-ear device.”), [0050] (“Typically, step d) includes assessing the acoustic seal of the in-ear device by subtracting said second sound level from said first sound level and using said transfer function and a compensation term relating to the in-ear device.”)

92. In my opinion, a person having ordinary skill in the art would have been motivated to apply *Voix*’s seal test signal frequency range of 100 Hz to 10,000 Hz to *Epley*’s audio system (as modified by *Padhi*’s cross-correlation teachings) for several reasons. First, while *Epley* discloses “applying a known sound” in seal assessment, *Epley* leaves the particular frequencies of the known sound to the person having ordinary skill in the art. *Epley* [0070].

Those of skill in the art will appreciate that microphone 76, amplifier 66 a/b and ADC 68 a/b that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ensure proper sealing of left and right ear probes. 18 a and 18 b. ***This is accomplished, for example, by applying a known sound*** and volume and measuring the acoustic response within the subject's ear as picked

up by the microphone. A straightforward comparison of input and output sound parameters determine whether the ear probe is properly sealed and further testing is avoided if the seal is not secure. The display produces a warning to the operator. Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

*Epley* [0070]. *Padhi* similarly does not specify a particular frequency. Thus, in my opinion, a person having ordinary skill in the art would have been motivated to look to other art—including *Voix*—to identify a suitable sound frequency to use for *Epley*'s acoustic seal test.

93. Second, *Voix* explains “sound typically having a frequency range varying from about 100 Hz to 10,000 Hz” produces “preferred octave bands which enclose most audible sounds.” *Voix* [0100]. Thus, a person having ordinary skill in the art would have recognized this frequency range would ensure the seal is tested for frequencies that impact user experience. A user's experience is most impacted by the introduction of unwanted sounds that the user can actually hear, and thus, testing specifically for whether those frequencies leak into a user's ear canal can best ensure that the seal is preventing entrance of those sound. In addition, *Epley* teaches its speaker emits audio in the frequency range “20 Hz and 8 kHz or perhaps even twice the upper limit.” *Epley*, [0074]. Twice 8 kHz is 16 kHz, which is 16,000 Hz. Accordingly, *Epley*'s speaker is capable of producing

sound within *Voix*'s test signal frequency range of 100 Hz to 10,000 Hz. Thus, in my opinion, a person having ordinary skill in the art would have found it straightforward to produce a known sound with a signal frequency range of "100 Hz to 10,000 Hz" as taught by *Voix*.

94. In my opinion, a person having ordinary skill in the art would have been motivated to apply *Voix*'s comparison of a value to a threshold when checking seal quality to the audio system taught by *Epley* in view of *Padhi* to determine when to send a "warning" to the user. *Epley* describes a "**display**" "produces a warning to the operator" in response to a "comparison" as part of a seal test. *Epley*, [0070]. But *Epley* but does not disclose what triggers this "warning." A person having ordinary skill in the art would have understood that the "warning" is a message to the operator that indicates that the seal needs to be adjusted. But, while the comparison of two signals would provide a value, the device needs some mechanism to check the value to determine if the value is indicative of a good seal or a bad seal.

95. Thus, a person having ordinary skill in the art would have been motivated to look to other art—including *Voix*—to determine when to elicit a "warning." For example, U.S. Patent No. 7,039,195 discloses sending a warning when a threshold is exceeded. EX1016, 10:17-20 ("The result of the analysis is compared to damage risk criteria and ***the user gets an audible or other form of***

**warning signal when certain limits are about to be exceeded** and actions have to be made.”). *Voix*’s teaching of the “presence of acoustic seal when attenuation...is equal to or larger than a predetermined threshold value” indicates a warning of low seal quality should be provided if the value is below the threshold value. *Voix*, [0039]. In this regard, a person having ordinary skill in the art would have understood that, in *Epley*’s seal assessment using cross-correlation as taught by *Padhi*, the cross-correlation is high when the microphone signal is very similar to the test signal produced in the user’s ear, which would occur when the seal highly attenuates the sound entering the ear. Thus, if the cross-correlation is low, based on *Voix*’s teaching, a person having ordinary skill in the art would have understood that a warning should be sent when the cross-correlation value is low.

96. A person having ordinary skill in the art would have further understood comparing a value (in particular, *Padhi*’s  $\alpha$ ) to a threshold would be predictably implemented, because *Padhi* already teaches that its value based on a cross-correlation calculation is compared to a threshold to produce a  $\text{Score}_1$ . See *Padhi*, [0078]. For example, *Padhi* shows that  $\alpha$  is compared to multiple thresholds (0.75, 0.5 and 0.25) to obtain  $\text{Score}_1$ .

TABLE 1

| Range                    | Score <sub>1</sub> |
|--------------------------|--------------------|
| $\alpha \geq 0.75$       | 60                 |
| $0.75 > \alpha \geq 0.5$ | 50                 |
| $0.5 > \alpha \geq 0.25$ | 40                 |
| $0.25 > \alpha$          | 30                 |

*Padhi*, [0078].

97. Further, in my opinion, a person having ordinary skill in the art would have been motivated to apply *Voix*'s "soft" "silicone" composition to the audio system taught by *Epley* in view of *Padhi* for *Epley*'s "ear-canal-easy-cushion." *Epley*, [0065]; *Voix*, [0099]. *Epley* teaches that its ear probes are "configured to be inserted into a subject's ear." *Epley*, [0004]. *Epley* further discloses "earpiece 70 is configured to have an ear-canal-easy cushion," but does not specify the material of the "cushion." *Epley*, [0065] ("Those of skill in the art will appreciate that earpiece 70 is configured to have an ear-canal-easy cushion, not shown, slipped over its distal end for the comfort of a test subject."). In my opinion, a person having ordinary skill in the art would therefore have been motivated to look to other art—including *Voix*—and would have recognized earpieces are "typically made out of generally soft materials such as silicones." *Voix*, [0099]. This would

have naturally led a person having ordinary skill in the art to implement *Epley*'s ear-canal-easy cushion using silicone.

98. For the reasons above, in my opinion a person having ordinary skill in the art would have had a reasonable expectation of success in applying *Voix*'s teachings of a test signal with a frequency ranging from 100-10,000 Hz, a threshold comparison to determine seal quality, and silicone ear tips to *Epley*'s device (using *Padhi*'s cross-correlation teachings) and would have known such a combination (yielding the claimed limitations) would predictably work and provide the expected functionality.

## 6. Claim 1

### a. [1.pre]: “An audio device configured to test a seal of an earphone comprising:”

99. I understand that “[a]n audio device configured to test a seal of an earphone comprising” is the preamble to claim 1 of the '493 patent. I have been asked to assume that the preamble is a claim limitation. It is my opinion that *Epley* discloses or suggests this feature.

100. *Epley*'s “sound and pressure generator and feedback mechanism 24” shown, for example, in Figs. 2 and 3, corresponds to the claimed “audio device configured to test a seal of an earphone.” *Epley*'s **feedback mechanism 24**

includes memory and a **microcontroller 54** that is configured to execute instructions to perform operations that test the seal of the ear terminal.

101. Epley's **feedback mechanism 24** and **ear probes** are illustrated in Figs. 2 and 3.

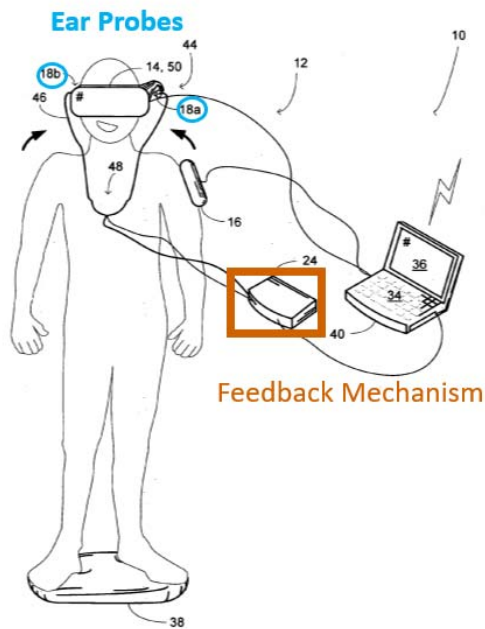


Fig. 2

Epley, Fig. 2

FIG. 3 is a detailed schematic diagram of sound and pressure generator and feedback mechanism 24.

Epley, [0061]

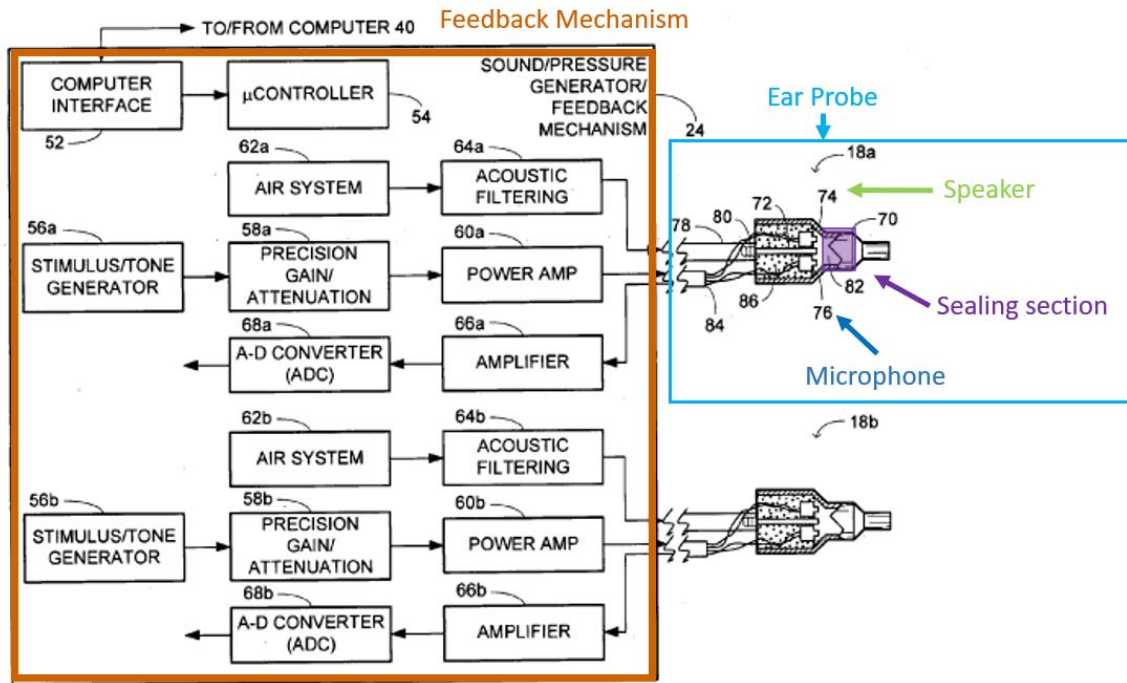


Fig. 3

Epley, Fig. 3.

102. Epley discloses that its **feedback mechanism** is coupled to “**ear probes 18**” by an umbilical cord that feeds sound to the ear probes. Epley, ¶[0051].

Those of skill in the art will appreciate that **ear probes 18** include air conduits and an umbilical cord feeding sound and/or pressure stimuli from **sound and pressure generator 24** under control of expert subsystem 26 operating within desk/laptop computer 40. It will be appreciated that computer 40 is shown in FIG. 2 as a laptop computer, but that computer 40 can take the alternative form of a PDA or other handheld device having a soft or hard keyboard 34 and a display screen 36. It will also be appreciated that **generator 24** can take the form of a small lightweight battery operated or USB operated portable device. In any event, those of skill in the art will appreciate that

lightweight portability is possible regardless of the form computer 40 takes, as carrying case 42 can take the form of a briefcase or computer tote bag facilitating the easy lifting or wheeling and thus transporting of system 10.

*Epley*, [0051].

103. *Epley* discloses that its **feedback mechanism** will “determine whether the **ear probe** is properly sealed” by “applying a known sound and volume and measuring the acoustic response.” *Epley*, ¶[0070].

Those of skill in the art will appreciate that microphone 76, amplifier 66 a/b and ADC 68 a/b that form part of the **feedback mechanism** are used in accordance with one embodiment of the invention to ensure proper sealing of **left and right ear probes. 18 a and 18 b**. This is accomplished, for example, by applying a known sound and volume and measuring the acoustic response within the subject's ear as picked up by the microphone. ***A straightforward comparison of input and output sound parameters determine whether the ear probe is properly sealed*** and further testing is avoided if the seal is not secure. The display produces a warning to the operator. Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

*Epley*, ¶[0070].

104. As I explain in more detail below, in connection with Claim 3, the claimed “audio device” is not limited to the “**sound and pressure generator and feedback mechanism 24**,” and may include “computer 40,” including “display screen 36.” *Epley*, [0029]

Those of skill in the art will appreciate that analyzers 20 and 22, expert subsystem 26, keyboard 34 and display 36 within the spirit and scope of the invention can take the form of a desktop or portable laptop computer 40, a personal digital assistant (PDA) or other small, lightweight, portable electronic device having at least data processing capability and preferably also having data displaying capability. Those of skill in the art also will appreciate that body-worn apparatus 12 includes a small, lightweight, portable headset 44 that at least partly integrates cameras 14 and ear probes 18 and one or more head/body sensors 16, as shown in FIG. 2. Indeed, either or both of the headset and sensor(s) can be battery operated. Those of skill in the art also will appreciate that **sound and pressure generator 24 can be battery operated and can be separate from computer 40 or integrated therein**, e.g. in the form of a custom input/output (I/O) card that plugs into a motherboard or I/O slot. Moreover, those of skill in the art will appreciate that destabilization device 38 broadly refers to any device particularly configured posturally to decrease the baseline stability of (i.e. to destabilize) a subject, e.g. it may take the form of a small step or incline or deformable footpad or cushion (e.g. a loosely fluid- or air-filled bladder, as shown) that, due at least in part to the subject's decreased baseline stability (e.g. balance shift) thereon tends to dis-

equilibriate the subject. Thus, preferably device 38 is also small, lightweight and portable.

*Epley*, [0029]

105. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Epley* to disclose or suggest a feedback mechanism that is configured to test a seal of the ear terminal (the claimed “audio device configured to test the seal of an earphone”).

**b. [1.1]: “a memory that stores instructions; and”**

106. In my opinion, *Epley* discloses or suggests “a memory that stores instructions.”

107. *Epley* discloses that **feedback mechanism 24** includes a **microcontroller 54** that uses “read-only memory (ROM)” to control operations of the components of the **feedback mechanism 24**. *Epley*, ¶[0061].

**Generator/feedback mechanism 24** includes a computer interface 52 and microcontroller (μCONTROLLER) 54 operatively coupled with one another. Computer interface 52 will be understood to be operatively coupled with computer 40. Those of skill will appreciate that **microcontroller 54** is programmed to provide control and monitoring interconnections with the remaining functional blocks to be described below, although for the sake of clarity no such operative interconnections are shown. For example, **microcontroller 54 can implement a state machine or other flow control mechanism that is**

hardwired, ***firmware-controlled (e.g. as by use of a read-only memory (ROM)*** or programmable logic array (PLA) or the like or software-controlled, e.g. using a writeable control store (WCS).  
***Microcontroller 54 can be as simple as a look-up table- or ROM-based controller that enforces operational parameters for the sound and pressure and feedback mechanisms.***

*Epley*, ¶[0061].

108. More generally, *Epley* further discloses that the functionality of each device—which a person of ordinary skill in the art would have understood to include the feedback mechanism 24 and its disclose seal test—is implemented by a “computer program in the form of instructions residing in a memory.” *Epley*, ¶[0099]

Finally, those of skill in the art will appreciate that ***the invented method, system and apparatus described and illustrated herein may be implemented in software, firmware or hardware***, or any suitable combination thereof. Preferably, the method system and apparatus are implemented in a combination of the three, for purposes of low cost and flexibility. Thus, ***those of skill in the art will appreciate that the method, system and apparatus of the invention may be implemented by a computer or microprocessor process in which instructions are executed, the instructions being stored for execution on a computer-readable medium and being executed by any suitable instruction processor.***

*Epley*, ¶[0099]

109. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Epley* to disclose or suggest that **feedback mechanism 24** includes ROM that stores instructions (“memory that stores instructions”).

**c. [1.2]: “a processor, wherein the processor is configured to execute the instructions to perform operations, the operations comprising:”**

110. In my opinion, *Epley* discloses or suggests “a processor, wherein the processor is configured to execute the instructions to perform operations, the operations comprising.”

111. *Epley* discloses that **feedback mechanism 24** includes “microcontroller 54 is programmed to provide control and monitoring interconnections with the remaining functional blocks” of the **feedback mechanism 24**, although such interconnections are not explicitly illustrated in the figures in *Epley*. *Epley*, ¶[0061].

**Generator/feedback mechanism 24** includes a computer interface 52 and **microcontroller (μCONTROLLER) 54** operatively coupled with one another. Computer interface 52 will be understood to be operatively coupled with computer 40. Those of skill will appreciate that **microcontroller 54** is programmed to provide control and monitoring interconnections with the remaining functional blocks to be described below, although for the sake of clarity no such operative interconnections are shown.

*Epley*, ¶[0061].

112. The **microcontroller 54** is illustrated as part of the feedback mechanism 24 in Fig. 3 of *Epley*.

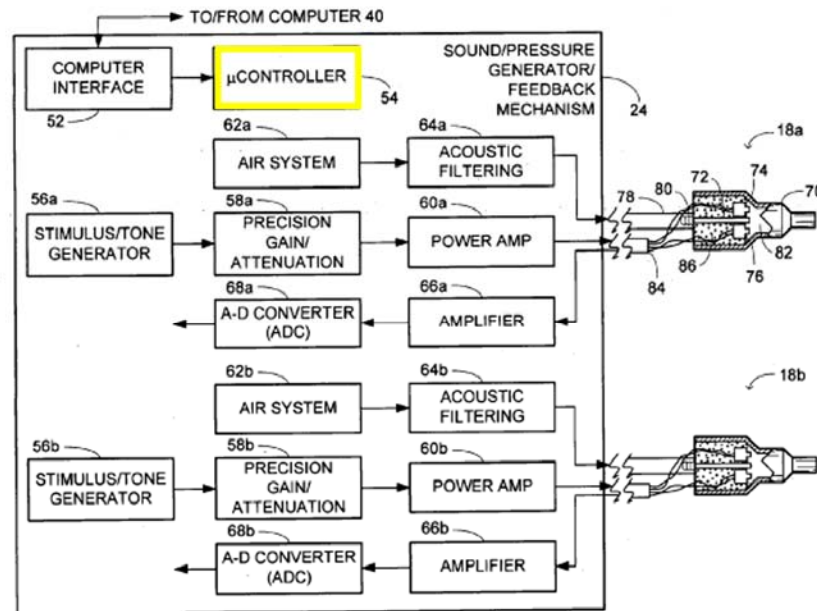


Fig. 3

*Epley*, Fig. 3.

113. Among the components controlled by the microcontroller 54 are the microphone 76, amplifier 66 a/b and ADC 68 a/b of feedback mechanism 24 for the purpose of the seal testing that is described below for the remaining claim limitations below. *Epley*, ¶[0070].

Those of skill in the art will appreciate that microphone 76, amplifier 66 a/b and ADC 68 a/b that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ensure proper sealing of left and right ear probes. 18 a and 18 b. This is

accomplished, for example, by applying a known sound and volume and measuring the acoustic response within the subject's ear as picked up by the microphone. A straightforward comparison of input and output sound parameters determine whether the ear probe is properly sealed and further testing is avoided if the seal is not secure. The display produces a warning to the operator. Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

*Epley*, ¶[0070].

114. And *Epley* explains that the functionality of each device-which a person of ordinary skill in the art would have understood to include the feedback mechanism 24 and its disclose seal test-is implemented by a “computer program in the form of instructions residing in a memory.” *Epley*, [0099]

Finally, those of skill in the art will appreciate that the invented method, system and apparatus described and illustrated herein may be implemented in software, firmware or hardware, or any suitable combination thereof. Preferably, the method system and apparatus are implemented in a combination of the three, for purposes of low cost and flexibility. Thus, those of skill in the art will appreciate that the method, system and apparatus of the invention may be implemented by a computer or microprocessor process in which instructions are executed, the instructions being stored for execution on a computer-

readable medium and being executed by any suitable instruction processor.

*Epley*, [0099].

115. Moreover, to the extent that “the processor is configured to execute the instructions to perform operations, the operations comprising” is a means-plus-function limitation, *Epley* discloses a processor, as explained above, and as explained below, the processor performed the algorithm disclosed by the ’493 patent.

116. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Epley* to disclose **feedback mechanism 24** includes microcontroller 54 (“processor”) that is programmed to provide control and monitoring interconnections with the remaining functional blocks of the **feedback mechanism 24** (“configured to execute the instructions to perform operations”).

**d. [1.2.1]: “generating a test signal,”**

117. In my opinion, *Epley* discloses or suggests “generating a test signal.”

118. As I explain above in connection with element [1.2], *Epley* discloses “instructions” “execut[ed] in a processor,” and those instructions control the operation of the feedback mechanism 24, including its seal test.

119. *Epley* discloses that the instructions are for “applying a known sound”/“output sound parameter[]” to “determine whether the ear probe is properly sealed.” *Epley*, [0070].

Those of skill in the art will appreciate that microphone 76, amplifier 66 a/b and ADC 68 a/b that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ensure proper sealing of left and right ear probes. 18 a and 18 b. This is accomplished, for example, by **applying a known sound and volume** and measuring the acoustic response within the subject's ear as picked up by the microphone. A straightforward comparison of input and output sound parameters determine whether the ear probe is properly sealed and further testing is avoided if the seal is not secure. The display produces a warning to the operator. Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

*Epley*, [0070].

120. Thus, as part of the feedback mechanism to ensure proper sealing, a “known sound” (test signal) is applied. A person of ordinary skill in the art would have understood that a “known sound” is a test signal, for the purpose of making a determination. And a person of ordinary skill in the art would have understood that

“applying” such a test signal requires actually generating the test signal. *Epley*, [0070].

121. Epley further discloses “stimulating one or more of a subject’s ears with sound...by executing in a suitable processor instructions that [are] stored in memory.” *Epley* [0093].

The method further includes a) at 604, ***stimulating one or more of a subject's ears with sound*** and/or pressure stimuli of a given characteristic; and b) at 606 periodically changing the given characteristic of the sound and/or pressure stimuli and c) at 608 repeating the stimulating step in accordance with the changed characteristic, thereby to prevent adaptation by the subject to the stimuli Steps 604 and 606 (along with the looping or repeating step 608) are performed while c) at 610 monitoring the subject for one or more of an ocular and sway response evoked by the given or changed stimuli. In accordance with one embodiment of the invention, ***at least the stimulating and the monitoring steps 604 and 606 (along with the looping or repeating step 608) are computer-assisted, e.g. accomplished by executing in a suitable processor instructions that stored in memory.*** In this manner, the invented method is at least semi-automated.

*Epley*, [0093]. By executing processor instructions, the subject’s ear(s) are stimulated with sound. While this disclosure of *Epley* is describing a test on a user, a person of ordinary skill in the art would have understood that other operations

involving the generation of sound and the analysis of that sound related to the operation of the feedback mechanism 24 are performed by the microcontroller 54 using instruction. A person of ordinary skill in the art would have thus understood (and at least found it obvious) that the processor executes instructions for generating the “known sound.”

122. Epley further discloses that the “stimulus/tone generator 56a” (which, as I explain above, is part of the feedback mechanism 24 controlled by the microcontroller 54) generates the known sound, and in particular, “programmed sound levels.” *Epley*, [0062]

Thus, a left channel sound and pressure generator and feedback mechanism includes ***stimulus/tone generator 56 a***, precision gain/attenuation device 58 a and power amp 60 a ***producing programmed sound levels***, tonal qualities and dynamics; air system 62 a and acoustic filtering device 64 a producing programmed pressure levels and dynamics; and amplifier 66 a and analog-to-digital (A-D) converter (ADC) 68 a producing a feedback signal representing sound and/or pressure and other conditions corresponding to the subject's left ear. Those of skill in the art will appreciate corresponding functional blocks 56 b, 58 b, 60 b, 62 b, 64 b, 66 b and 68 b identically produce programmed pressure levels and dynamics, programmed sound levels, tonal qualities and dynamics and a feedback signal representing sound and/or pressure and other conditions corresponding to the subject's right ear. ***Stimulus/tone***

***generator 56 a, precision gain/attenuation device 58 a and power amp 60 a will be referred to herein collectively as a sound generator.***

Air system 62 a and acoustic filtering device 64 a will be referred to herein collectively as an air pressure generator.

*Epley*, [0062].

123. A person of ordinary skill in the art would have understood from these disclosures that the microcontroller generates the known sound based on “suitable processor instructions that [are] stored in memory,” and programs the sound generator with the known sound, and the sound generator in turn sends the “known sound” to the ear probe.

124. Moreover, to the extent the “processor is configured to execute the instructions to perform operations, the operations comprising” limitation is determined to be a means-plus-function limitation, *Epley*’s disclosure is the same or equivalent to the algorithm disclosed in the ’493 patent at step 304, which requires “A test signal is acquired in a step 304. For example, the test signal can be stored in memory or generated by a microprocessor.” ’493 patent, 8:6-8.

125. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Epley* to disclose or suggest instructions executed in the processor for generating a test signal.

**(a) [1.2.1.1]: “wherein the test signal includes multiple frequency components, wherein**

**at least one component lies between a range of 30 Hz and 200 Hz;”**

126. In my opinion, Voix discloses or suggests “wherein the test signal includes multiple frequency components, wherein at least one component lies between a range of 30 Hz and 200 Hz.”

127. First, Voix discloses “a method for assessing in-situ an acoustic seal of an in-ear device” that includes “measuring a first sound level.” Voix, [0036].

In accordance with an aspect of the present invention, there is provided ***a method for assessing in-situ an acoustic seal of an in-ear device*** using an apparatus having a sound measurement device connected thereto, the in-ear device having a passageway extending therethrough, the passageway being in fluid communication with an ear canal of an ear when the in-ear device is inserted therein, the sound measurement device being removably engageable into the passageway, said method comprises: a) calibrating said sound measurement device by measuring a reference sound level with said sound measurement device when being submitted to a reference sound source and when being located in a close relationship relative thereto; b) ***measuring a first sound level*** outside the ear canal with said sound measurement device ***when submitted to the sound source*** and when located in a close relationship relative to the in-ear device and outside the ear canal; c) measuring a second sound level inside the ear canal with said sound measurement device when submitted to the sound source and when engaged into and occluding the passageway with the in-ear device inserted inside the ear canal; and d) assessing the



operation thereof. ***The sound source 24 provides a sound typically having a frequency range varying from about 100 Hz to about 10000 Hz*** such that the following preferred octave bands which enclose most audible sounds (including conversation frequency range of about 300 Hz to about 3000 Hz) are essentially covered: 125 Hz, 250 Hz, 500 Hz, 1000 Hz, 2000 Hz, 4000 Hz and 8000 Hz. Typically, the controller 22 is a personal-type computer, a laptop, a palm computer or the like that include a central processing unit (not shown) or the like to process the data and perform assessments. A dual microphone probe 26 that is removably engageable into the passageway 20 of the earplug 12 operatively connects to the controller 22. The probe 26 typically includes first 28 and second 30 sound measurement devices or microphones mounted in a back-to-back relationship relative to one another such that the first microphone 28 measures an external sound pressure level in proximity to the earplug 12 and the second microphone 30 measures an internal sound pressure level inside the ear canal 14 when the earplug 12 is inserted therein, respectively.

Voix, [0100].

130. It is my opinion that a person of ordinary skill in the art would have understood that a sound with a “frequency range varying from about 100 Hz to about 10000 Hz” includes multiple frequency components spread across the entire range, including multiple components in the 30 Hz to 200 Hz range. A frequency component of a signal is how much of that frequency is in the signal. Where Voix

refers to a “frequency range” that extends between 100 to 10000 Hz, this means that the produced signal includes frequency components that span this range of frequencies. This is consistent with the ’493 patent disclosure of multiple frequency components across a frequency range:

In one exemplary embodiment, filter step 1306 filters the audio signal from 50 Hz to 150 Hz which corresponds to ***a frequency range of the test signal.***

’493 patent, 15:43-46.

131. As I described in §IX.A.5, it is my opinion that a person of ordinary skill in the art would have been motivated to apply ***Voix’s*** teachings of a test signal range to ***Epley’s*** feedback mechanism.

132. Accordingly, it is my opinion that a person having ordinary skill in the art would have understood that Voix discloses a test signal with components that range between 20 Hz and 10,000 Hz (“the test signal includes multiple frequency components, wherein at least one component lies between a range of 30 Hz and 200 Hz”).

- (ii) [1.2.2]-[1.2.3]: “sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user,”

133. In my opinion, *Epley* discloses or suggests “sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user.”

134. As I explain above in connection with element [1.2], *Epley* discloses “instructions” “execut[ed] in a processor,” and those instructions control the operation of the feedback mechanism 24, including its seal test.

135. Moreover, *Epley* discloses “applying a known sound” (or “output sound parameter”) to “ear probe 18a.” *Epley*, [0070].

Those of skill in the art will appreciate that microphone 76, amplifier 66 a/b and ADC 68 a/b that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ***ensure proper sealing of left and right ear probes. 18 a and 18 b***. This is accomplished, for example, ***by applying a known sound and volume*** and measuring the acoustic response within the subject's ear as picked up by the microphone. ***A straightforward comparison of*** input and ***output sound parameters*** determine whether the ear probe is properly sealed and further testing is avoided if the seal is not secure. The display produces a warning to the operator. Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

*Epley*, [0070].

136. And each ear probe includes a “speaker 74,” coupled to the feedback mechanism 24 via a wiring harness 84. Moreover, the speaker 74 has a conduit to the ear canal such that *Epley*, [0064].

***Left ear probe 18 a includes*** a tapered, generally cylindrical earpiece 70 containing an air tube 72, ***a miniature speaker (referred to in the audiology literature as a “receiver”) 74*** and an optional microphone 76. Air tube 72 is securely connected to an air hose 78 via an irregularly shaped, interference fitted connector 80. The distal end of air tube 72 extends to a hollow distal region 82 of earpiece 70, as shown. The proximal end of air tube 72 is in fluid (air) communication via air hose 78 with acoustic filtering device 64 a/b. ***Speaker 74*** and microphone 76 are respectively ***connected to power amp 60 a/b*** and amplifier 66 a/b ***via a wiring harness 84*** containing at least two signal wire pairs, e.g. twisted signal wire pairs, as shown (although those of skill in the art will appreciate that some microphones provide a third bias terminal and thus require three wires instead of two). ***The distal ends or so-called ‘snouts’ of speaker 74 and microphone 76 also extend to hollow distal region 82 of earpiece 70, as shown.***

*Epley*, Fig. 3.

137. A person of ordinary skill in the art would have understood (and at least found it obvious) that in order to apply a “known sound” such that the

“acoustic response” is “picked up by the microphone,” the signal needs to be played by “speaker 74” of “ear probe 18a,” based on instructions in the processor. For example, *Epley* further discloses “stimulating one or more of a subject’s ears with sound...by executing in a suitable processor instructions that [are] stored in memory.” *Epley* [0093]. A person of ordinary skill in the art would have understood from these disclosures that the microcontroller generates the known sound based on “suitable processor instructions that [are] stored in memory,” and programs the sound generator with the known sound, through which the “known sound” is sent to the ear probe.

138. Moreover, to the extent the “processor is configured to execute the instructions to perform operations, the operations comprising” limitation is determined to be a means-plus-function limitation, *Epley*’s disclosure is the same or equivalent to the algorithm disclosed in the ’493 patent at step 306, which requires “The test signal is provided to the acoustic transducer.” ’493 patent, 8:9-10.

139. Second, *Epley* discloses that the ear terminal includes the claimed “first microphone,” “speaker,” and “sealing section.”

140. In particular, *Epley* discloses “ear probe 18[a/b]” includes a first “microphone 76,” the “speaker 74,” and “earpiece 70,” where “earpiece 70 is

configured to have an ear-canal-easy cushion.” *Epley*, [0065]. These structures are shown in Figure 3.

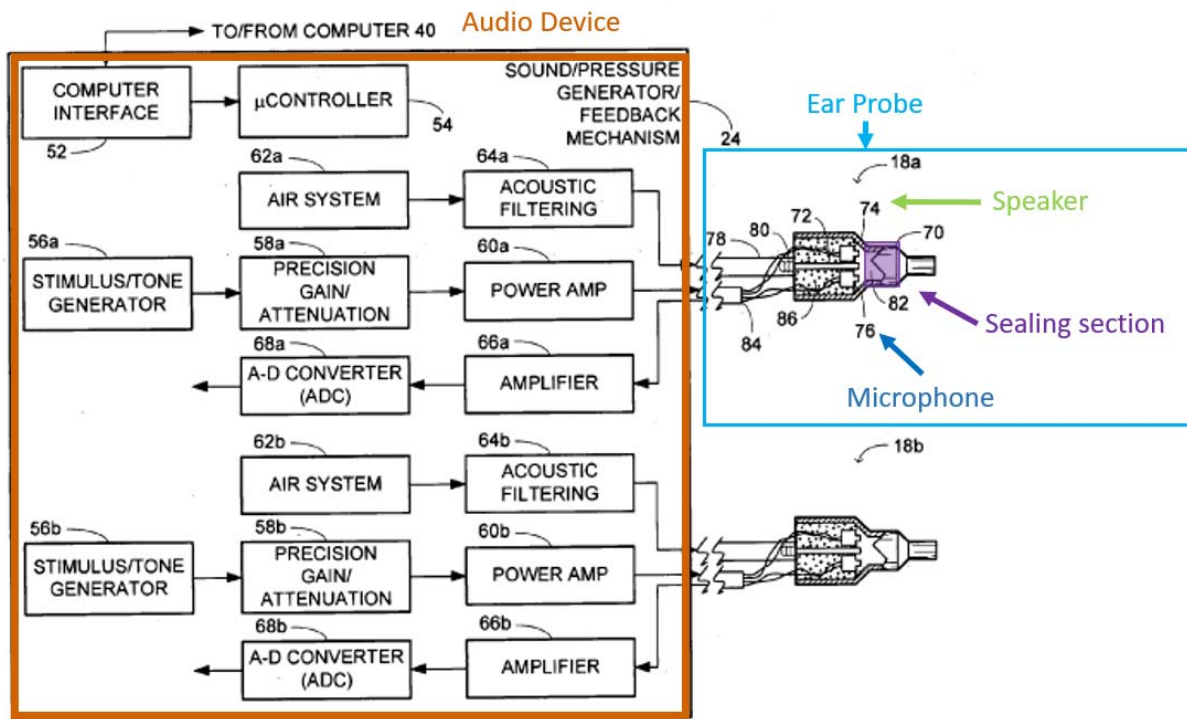


Fig. 3

*Epley*, Fig. 3.

In accordance with one embodiment of the invention, air tube 72, **speaker 74 and microphone 76 are secured within earpiece 70 by potting compound 86**, e.g. of epoxy resin. In accordance with one embodiment of the invention, potting compound 86 extends only partway toward the distal end of earpiece 70, thus forming hollow (air-filled) distal region 82 that is in fluid (air) communication with the subject's left ear. Alternatively, yet within the spirit and scope of the invention, the potting compound can be of any suitable material that instead may fill the entire interior region of earpiece 70. In

accordance with one embodiment of the invention, earpiece 70 is of machined or extruded aluminum, although within the spirit and scope of the invention it can be of any suitable material. ***Those of skill in the art will appreciate that earpiece 70 is configured to have an ear-canal-easy cushion, not shown, slipped over its distal end for the comfort of a test subject.***

*Epley*, [0065].

141. The probe with earpiece 70 “is configured to be inserted into a subject’s ear in an operational alignment with the subject’s ear canal and sealing engagement with the wall of the subject’s ear canal.” *Epley*, [0004], [0017] (***“[E]ar probes sealingly engage the subject’s right and/or left ear canal walls and operate to convey aural pressure or sound thereto, as will be seen.”***). This is to provide “continuously sealed acoustic engagement with the subject’s ear canals,” as confirmed by the shape of earpiece 70 as illustrated in Fig. 3. To the extent other components of ear probe 18a are also involved in sealing (e.g., the cushion over the distal end of the earpiece 70 that is not shown in Figure 3 (see *Epley*, [0065])), they are also part of the claimed “sealing section.” This acoustic sealing engagement with the ear canal walls creates a barrier between inside and outside of the ear, *i.e.*, between two sides of the ear probe 18[a/b].

Those of skill in the art also will appreciate that providing bilateral ear stimulus devices within the headgear that forms a part of the invented

system and apparatus makes testing more repeatable and standardized because *the stimulus devices automatically, reliably (and sealingly, in the case of the insertable ear probes 18 described and illustrated herein) seat themselves within the subject's corresponding ear canals once and for the duration of a complete test session.*

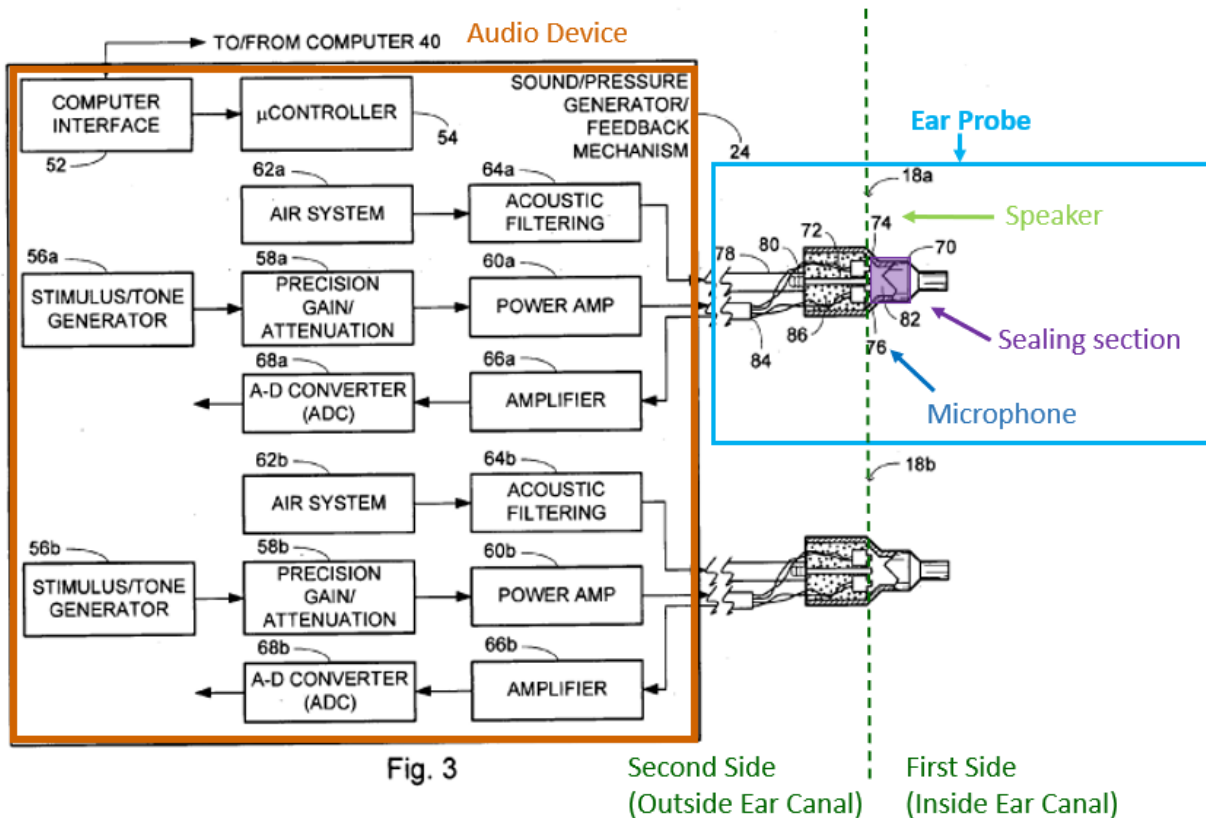
Epley, [0042]

142. Moreover, use of a “known sound” to test the seal only works if the sealing engagement creates an “acoustic” barrier. That is, if the ear probes did not provide an acoustic barrier, sound would easily flow in and out of the ear, and the seal test using a known sound would not be able to determine the quality of seal.

Those of skill in the art will appreciate that microphone 76, amplifier 66 a/b and ADC 68 a/b that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ***ensure proper sealing of left and right ear probes. 18 a and 18 b. This is accomplished, for example, by applying a known sound*** and volume and measuring the acoustic response within the subject's ear as picked up by the microphone. A straightforward comparison of input and output sound parameters determine whether the ear probe is properly sealed and further testing is avoided if the seal is not secure. The display produces a warning to the operator. Thus a first use for this circuitry is ***during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.***

*Epley*, [0070].

143. The acoustic barrier is between the inside of the ear canal and outside the ear canal, as illustrated below:



144. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Epley* to disclose or suggest that its processor execute instructions stored in the ROM to send the known sound to **speaker 74** of the ear probe (“sending the test signal to a speaker of an earphone”). Further, in my opinion, a person having ordinary skill in the art would have understood *Epley* to disclose or suggest that the ear probe includes and **microphone 76** (“first

microphone”), **speaker 74** (“speaker”), and **sealing section** (“sealing section”).

Finally, in my opinion, a person having ordinary skill in the art would have understood *Epley* to disclose or suggest that *Epley*’s ear terminal includes a **sealing section** that forms an acoustic sealing engagement with the ear canal to form a barrier between the inside of the ear canal and outside of the ear canal (“wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user”).

**(iii) [1.2.4]: “receiving a first microphone signal from the first microphone;”**

145. In my opinion, *Epley* discloses or suggests “receiving a first microphone signal from the first microphone.”

146. As I explained in [1.2], *Epley* discloses “instructions” that are “execut[ed] in a processor,” and those instructions control the operation of the feedback mechanism 24, including its seal test. *Epley*, [0099]. Such instructions include instructions to receive the signal from the **microphone 76**, which is “connected” to the microcontroller 54 via “wire pairs,” amplifier 66a and ADC 68a.

147. The path from the **microphone 76** through amplifier 66a to the ADC 68a is illustrated in Figure 3.

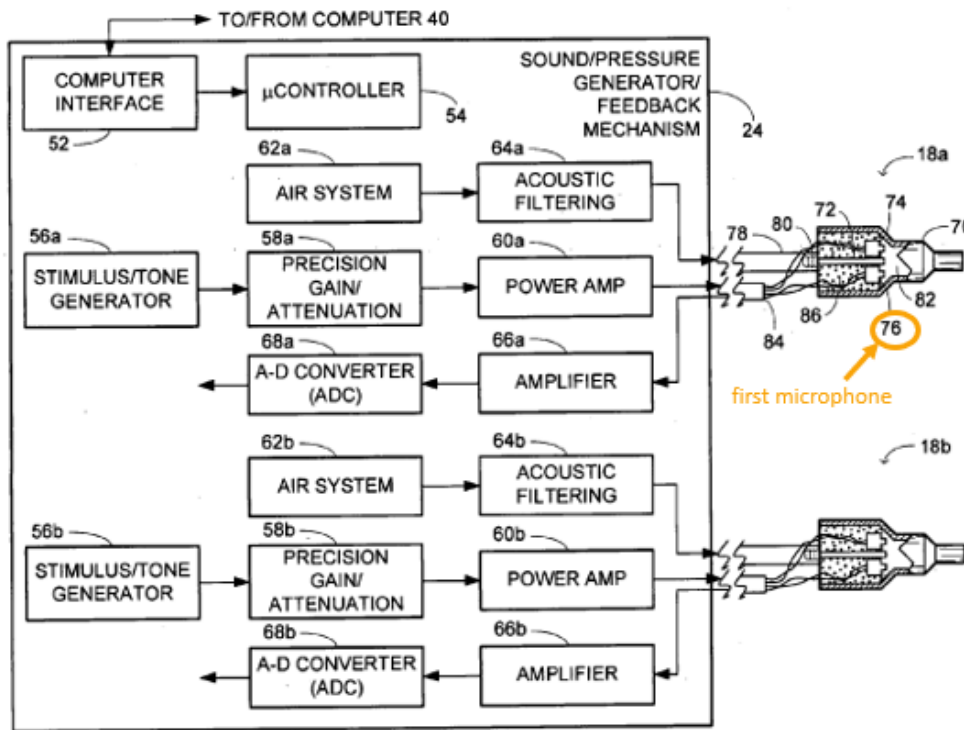


Fig. 3

Epley, Fig. 3

148. The microphone 76 is connected to amplifier 66a/b via wire pairs.

Epley, [0064].

Left ear probe 18 a includes a tapered, generally cylindrical earpiece 70 containing an air tube 72, a miniature speaker (referred to in the audiology literature as a “receiver”) 74 and an optional microphone 76. Air tube 72 is securely connected to an air hose 78 via an irregularly shaped, interference fitted connector 80. The distal end of air tube 72 extends to a hollow distal region 82 of earpiece 70, as shown. The proximal end of air tube 72 is in fluid (air) communication via air hose 78 with acoustic filtering device 64 a/b. **Speaker 74 and microphone 76 are respectively**

*connected to power amp 60 a/b and amplifier 66 a/b via a wiring harness 84 containing at least two signal wire pairs*, e.g. twisted signal wire pairs, as shown (although those of skill in the art will appreciate that some microphones provide a third bias terminal and thus require three wires instead of two). The distal ends or so-called ‘snouts’ of speaker 74 and microphone 76 also extend to hollow distal region 82 of earpiece 70, as shown.

*Epley*, [0064].

149. Further, as I explain above in connection with element [1.2], the microcontroller 54 is coupled to other components in the feedback mechanism 24, including the ADC 68a to receive the microphone signal. *Epley*, ¶[0061].

150. Further, such instructions include to cause the microphone 76 to “measur[e] the acoustic response within the subjects ear as picked up by the microphone.” *Epley*, [0070].

Those of skill in the art will appreciate that *microphone 76, amplifier 66 a/b and ADC 68 a/b* that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ensure proper sealing of left and right ear probes. 18 *a* and 18 *b*. This is accomplished, for example, by applying a known sound and volume and *measuring the acoustic response within the subject's ear as picked up by the microphone*. A straightforward comparison of *input* and output *sound parameters* determine whether the ear probe is properly sealed and further testing

is avoided if the seal is not secure. The display produces a warning to the operator. Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

151. Moreover, to the extent the “processor is configured to execute the instructions to perform operations, the operations comprising” limitation is determined to be a means-plus-function limitation, *Epley*’s disclosure is the same or equivalent to the algorithm disclosed in the ’493 patent, which discloses “ECM 106 is configured to measure the test signal in ear canal 124 and provide the measured test signal to processor 116.” ’493 patent, 8:58-60.

152. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Epley* to disclose or suggest that its processor execute instructions stored in the ROM to measure the acoustic response within the subject’s ear as picked by the microphone 76 (“receiving a first microphone signal from the first microphone”).

- (iv) **[1.2.5]: “calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination;”**

153. In my opinion, *Epley* in view of *Padhi* discloses or suggests “calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination.”

154. This limitation requires “calculating a value,” where the calculation uses “a first signal and a second signal.” I note that claim 1 does not specify what the recited “first signal” and “second signal” are or what generated them. Regardless, *Epley* in view of *Padhi* discloses calculating a “value” using a “first signal” and a “second signal.”

155. As I explain above in connection with element [1.2], *Epley* discloses “instructions” “execut[ed] in a processor,” and those instructions control the operation of the feedback mechanism 24, including its seal test.

156. As part of this operation of the feedback mechanism 24, *Epley* discloses a “comparison” of the “known sound” and the “acoustic response” that is picked up by the microphone. *Epley*, [0070].

Those of skill in the art will appreciate that microphone 76, amplifier 66 *a/b* and ADC 68 *a/b* that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ensure proper sealing of left and right ear probes. 18 *a* and 18 *b*. This is accomplished, for example, ***by applying a known sound and volume and measuring the acoustic***

*response within the subject's ear as picked up by the microphone. A straightforward comparison of input and output sound parameters determine whether the ear probe is properly sealed* and further testing is avoided if the seal is not secure. The display produces a warning to the operator. Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

Epley, [0070].

157. As I note above in section IX.A.3, *Epley* does not provide details for how to compare the known sound and the acoustic response.

158. *Padhi* provides details for how to calculate a value using cross correlation of a first signal and a second signal to determine the similarity between the two signals, including how much the signals “match.” *Padhi*, [0017].

In one aspect of operation, the apparatus 100 receives audio signals, such as an input signal 102 and a reference signal 104. ***The apparatus 100 cross-correlates the audio signals and uses the cross correlation to identify various characteristics or parameters of the audio signals.*** For example, the cross correlation may be used to identify a variation of pitch between the signals 102, 104. The cross correlation may also be used to identify a delay between the signals 102, 104. ***In addition, the cross correlation may be used to identify an amount of correlation between the signals 102, 104. Using the identified***

*characteristics, the apparatus 100 generates a score or other indicator identifying the extent to which the signals 102, 104 match.*

*Padhi*, [0017].

The apparatus 100 uses the identified parameters to generate a score at step 610. *The score identifies the extent to which the signals 102, 104 match.* This may include, for example, the score generator 144 receiving the identified parameters from the identifiers 138-142 and generating the score using the parameters.

*Padhi*, [0074].

159. *Padhi's* Fig.1 illustrates an apparatus that receives an **input audio signal 102** via a microphone. *Padhi*, [0018]. A **reference signal 104** is supplied by a reference signal source. *Id.*

In the illustrated example, the apparatus 100 receives an **input signal 102** and a **reference signal 104**. The **input signal 102** and the **reference signal 104** represent audio signals to be compared by the apparatus 100. In this example, the **input signal 102** is supplied by a microphone 106, and the **reference signal 104** is supplied by a reference signal source 108. In a particular embodiment, the **input signal 102** is produced by a microphone 106 used by a singer, and the **reference signal 104** represents an original song provided by a compact disc (CD) player. In another particular embodiment, the **input signal 102** is produced by a microphone 106 used by a student, and the **reference signal 104** represents a proper pronunciation of

words to be spoken by the student. This represents two possible uses of the apparatus 100. **However, the illustrated input signal 102 and reference signal 104 could be replaced by any suitable audio signals provided by any suitable source or sources.**

Padhi, [0018].

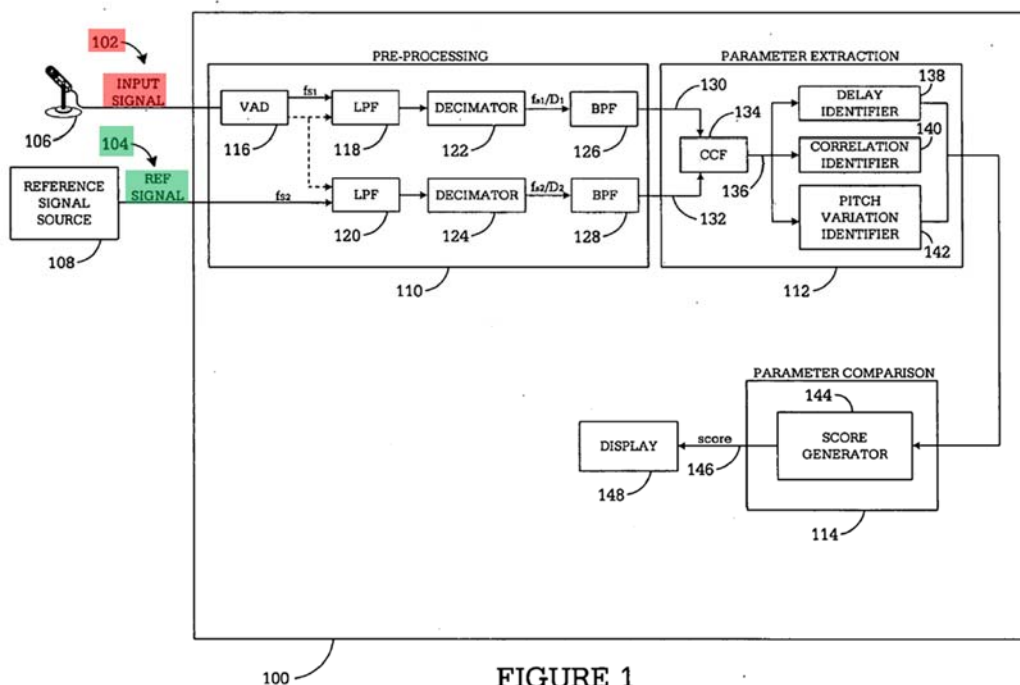


FIGURE 1

Padhi, Fig. 1.

160. At step 604, “apparatus 100 filters and decimates the input and reference signals.” *Id.*, [0071]. “This may include ... filters 118, 120 performing low pass filtering” on the **input signal 102** (*Epley*’s acoustic response within the subject’s ear as picked by the microphone 76) and the **reference signal 104** (*Epley*’s known sound). *Id.* At step 606, “apparatus 100 cross-correlates the

processed signals.” *Id.*, [0072]. “This may include ... the **cross correlator 134** correlating the pre-processed input and reference signals 130, 132 to produce a cross-correlated signal 136.” *Id.* At step 608, “apparatus 100 identifies a plurality of parameters using the cross-correlated signals.” *Id.*, [0073]. “This may include ... the identifiers 138-142 using the cross-correlated signal 136 to identify a delay between the signals 130, 132, an amount of correlation between the signals 130, 132, and a pitch variation between the signals 130, 132.” *Id.*

The apparatus 100 filters and decimates the input and reference signals at step 604. This may include, for example, the filters 118, 120 performing low pass filtering on the signals **102**, **104**. This may also include the decimators 122, 124 decimating the filtered input signals **102**, **104**. This may further include the filters 126, 128 band pass filtering the decimated input signals **102**, **104**, which produces two pre-processed signals 130, 132.

*Padhi*, [0071].

The apparatus 100 cross-correlates the processed signals at step 606. This may include, for example, the **cross correlator 134** correlating the pre-processed input and reference signals 130, 132 to produce a cross-correlated signal 136. The **cross correlator 134** may correlate an equal number of samples from the pre-processed signals 130, 132. The **cross correlator 134** could also correlate an unequal number of samples from the pre-processed signals 130, 132.

*Padhi*, [0072].

The apparatus 100 identifies a plurality of parameters using the cross-correlated signals at step 608. This may include, for example, the identifiers 138-142 using the cross-correlated signal 136 to identify a delay between the signals 130, 132, an amount of correlation between the signals 130, 132, and a pitch variation between the signals 130, 132. Other or additional parameters could also be identified.

*Padhi*, [0073].

161. At [0077]-[0082], *Padhi* describes how score generator 144 generates a score. “First, the score generator 144 generates a value  $\alpha$  using information from the delay identifier 138 and the correlation identifier 140.” *Id.*, [0077]. The value  $\alpha$  is determined according to the formula  $\text{local\_maxout}/\text{max\_out}$ . *Id.*

“[L]ocal\_maxout identifies the largest value of the cross-correlated signal within a certain range...” *Id.*, [0078]. Further, “max\_out identifies the largest value in the cross-correlated signal.” *Id.*

The following represents one example way in which the score generator 144 may generate a score 146. Various values used in this example are for illustration and explanation only. First, the score generator 144 generates a value  $\alpha$  using the information from the delay identifier 138 and the correlation identifier 140. The value of  $\alpha$  is determined using the formula:

$$\text{local\_maxout}/\text{max\_out}.$$

*Padhi*, [0077].

In this embodiment, `local_maxout` identifies the largest value of the cross-correlated signal within a certain range centered at the 128th lag, such as within a range of 125-131 ( $\pm 3$  lags from 128). Also, `max_out` identifies the largest value in the cross-correlated signal. In this example, the correlation identifier 140 outputs two values, one identifying a value for `max_out` and another identifying a value of `local_max_out`. Using the value of  $\alpha$ , the score generator 144 selects a first score ( $\text{Score}_1$ ) using Table 1.

*Padhi*, [0078].

162. In my opinion, regardless of how this limitation is interpreted, *Padhi*'s cross-correlation teaches or at minimum renders obvious a non-phase calculation when performed on the multi-frequency component test signal and microphone signal as taught by *Epley* in view of *Voix*. As I explain in §IX.A.6.d(a), the test signal would have multiple frequency components ranging between 100 Hz and 10000 Hz. I have been asked to consider the scenario where Patent Owner takes the position cross-correlation is a non-phase calculation. In that case, *Padhi* discloses this for the reasons I discuss above in this section. I have also been asked to consider the scenario where the non-phase calculation is interpreted to require a calculation performed on two signals without phase. The notion of phase applies to sinusoidal signals and it is a single value that indicates when the signal “starts”

relative to another signal. For example, a cosine signal is phase shifted by  $90^\circ$  relative to a sine signal. This notion of phase does not apply to more complex signals such as a pseudorandom sequence. Again, if the non-phase calculation is interpreted to require a calculation performed on two signals without phase, this is disclosed or at least rendered obvious. For example, Epley leaves it to a POSITA to determine what “known signal” to use, and one example of a known test signal was a “pseudorandom sequence,” which a POSITA would have understood to not have a phase and to be included as a “known signal.” This well-known feature was taught, *e.g.*, by *Svean*, which explains that its embodiment in Figure 8 is for performing a seal test. *Svean*, 12:51-13:5. *Svean* further explains that a “pseudorandom sequence” can be used for a seal test. *Svean*, 12:38-48. A pseudorandom sequence is just a signal represented by a randomly generated series of numbers in time, and therefore it does not have a phase. Further, because the microphone signal includes the pseudorandom test signal and random background sound, it also has no phase (since neither signal is a sinusoidal signal). A POSITA would have been motivated to use such a test signal because this was one known technique of implementing a known sound. Moreover, testing with a pseudorandom sequence is advantageously immune to background noise because background noise is not correlated to the pseudorandom sequence test signal.

Thus, as taught by Svean, the use of a pseudorandom test signal would have yielded predictable results.

163. As I explain above §IX.A.3, a person of ordinary skill in the art would have been motivated to apply *Padhi*'s teachings of comparing audio signals through cross correlation to *Epley*'s seal analyzer (feedback mechanism 24).

164. Moreover, to the extent the “processor is configured to execute the instructions to perform operations, the operations comprising” limitation is determined to be a means-plus-function limitation, *Padhi*'s disclosure is the same or equivalent to the algorithm disclosed in the '493 patent at step 310, which requires “[d]etermin[ing] degree of acoustic sealing via cross-correlation of test signal with ECM signal.” '493 patent, Fig. 3.

165. Accordingly, in my opinion, a person having ordinary skill in the art would have understood that *Epley* in view of *Padhi* to disclose or suggest that instructions executed in the processor are configured to execute instructions stored in the memory to calculate a value that is the result of a non-phase cross correlation calculation ( $\alpha$ ) with the input signal and reference signal (“calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination”).

(v) [1.2.6]: “comparing the value to a threshold to determine a seal quality between the first side and the second side;”

166. In my opinion, *Epley* in view of *Padhi* and *Voix* teaches or suggests “comparing the value to a threshold to determine a seal quality between the first side and the second side.”

167. As I explain above in connection with element [1.2], *Epley* discloses “instructions” “execut[ed] in a processor,” and those instructions control the operation of the feedback mechanism 24, including its seal test. Such instructions include instructions for “determin[ing] whether the ear probe is properly sealed.” *Epley*, [0070].

Those of skill in the art will appreciate that microphone 76, amplifier 66 a/b and ADC 68 a/b that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ensure proper sealing of left and right ear probes. 18 a and 18 b. This is accomplished, for example, by applying a known sound and volume and measuring the acoustic response within the subject's ear as picked up by the microphone. A straightforward comparison of input and output sound parameters **determine whether the ear probe is properly sealed** and further testing is avoided if the seal is not secure. The display produces a warning to the operator. Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

*Epley*, [0070]. As I explain above in connection with element [1.2.3], the sealing is with respect to the first and second sides (inside and outside the ear canal).

168. *Voix* discloses comparing the “attenuation between the first and second sound levels” to a “predetermined threshold value,” for purposes of an “acoustic seal test.” [0140]

Step d) further includes ***indicating presence of acoustic seal*** when ***attenuation between the first and second sound levels*** is equal to or larger than ***a predetermined threshold value*** within a predetermined frequency range. As an example, the predetermined threshold value could be 11 dB (the NR, ANSI S3.19 test, within the 98% percentile) at about 250 Hz.

*Voix*, [0140].

169. The first and second sound levels are collected from “microphones mounted in a back-to-back relationship relative to one another such that the first microphone 28 measures an external sound pressure level in proximity to the earplug 12 and the second microphone 30 measures an internal sound pressure level inside the ear canal 14.” [0100]

The apparatus 10 typically includes a controller unit 22 operatively connected to a broadband reference sound source 24 to control operation thereof. The sound source 24 provides a sound typically having a frequency range varying from about 100 Hz to about 10000

Hz such that the following preferred octave bands which enclose most audible sounds (including conversation frequency range of about 300 Hz to about 3000 Hz) are essentially covered: 125 Hz, 250 Hz, 500 Hz, 1000 Hz, 2000 Hz, 4000 Hz and 8000 Hz. Typically, the controller 22 is a personal-type computer, a laptop, a palm computer or the like that include a central processing unit (not shown) or the like to process the data and perform assessments. A dual microphone probe 26 that is removably engageable into the passageway 20 of the earplug 12 operatively connects to the controller 22. The probe 26 typically includes first 28 and second 30 sound measurement devices or ***microphones mounted in a back-to-back relationship relative to one another such that the first microphone 28 measures an external sound pressure level in proximity to the earplug 12 and the second microphone 30 measures an internal sound pressure level inside the ear canal 14*** when the earplug 12 is inserted therein, respectively.

Voix, [0100]. Figure 1 illustrates the first and second sound levels measured by the first microphone 28, and second microphone 30.



170. Voix does this for the purpose of “an acoustic seal test.” Voix, Abstract; see also [0036].

171. Voix determines whether the attenuation between the first and second sound levels is “equal to or larger than a predetermined threshold value.” Voix, [0039].

Typically, step d) includes indicating presence of acoustic seal when ***attenuation between said first and second sound levels is equal to or larger than a predetermined threshold value*** within a predetermined frequency range.

Voix, [0039].

Step d) further includes ***indicating presence of acoustic seal when attenuation between the first and second sound levels is equal to or larger than a predetermined threshold value*** within a predetermined frequency range. As an example, the predetermined threshold value could be 11 dB (the NR, ANSI S3.19 test, within the 98% percentile) at about 250 Hz.

Voix, [0140].

172. As I explain in §IX.A.5, it is my opinion that a person of ordinary skill in the art would have been motivated to apply Voix’s teachings of using a threshold to assess the seal quality to Epley’s feedback mechanism.

173. In my opinion, in *Epley* in view of *Padhi* and *Voix*, comparison to a threshold would demonstrate the level of similarity between the known sound and acoustic response. In other words, a calculated cross correlation that is above a threshold would indicate that the two signals are similar. Thus, in my opinion, a person having ordinary skill in the art reviewing the teachings of *Epley*, *Padhi* and *Voix* would have understood that comparison of the calculated value to a threshold indicates whether these two signals are sufficiently similar such that there is a good seal quality.

174. To the extent that the Board finds “seal quality” is not binary in nature, it would have been obvious to use a non-binary seal quality to provide more information to a user. For example, U.S. Patent 6,671,643 discloses using “tone combinations” to indicate the test result for a hearing aid. *See, e.g.*, EX1019, 2:45-51 (“To provide a conclusion as to the result of the test, the hearing aid emits a monitoring signal...produced in audible form, for example by emitting specific tones or tone combinations which allow a conclusion about the result of the test”). The use of “tone combinations”—as opposed to a single tone—indicates multiple values as opposed to a result with a binary nature.

175. Moreover, to the extent the “processor is configured to execute the instructions to perform operations, the operations comprising” limitation is determined to be a means-plus-function limitation, *Voix*’s comparison of a value to

a threshold as applied to *Epley*'s seal analyzer using *Padhi*'s cross-correlation teachings such that a value based on the cross-correlation (like *Padhi*'s  $\alpha$ ) is compared to a threshold is the same or equivalent to the algorithm disclosed in the '493 patent at step 416, which requires "[t]he instantaneous cross-correlation (i.e. the cross-correlation at zero-lag) value from step 412 is compared with the cross-correlation threshold value 414 using comparator unit 416." '493 patent, 9:45-48.

176. Accordingly, in my opinion, a person having ordinary skill in the art would have understood that *Epley* in view of *Padhi* and *Voix* discloses or suggests that instructions executed in a processor and stored in a memory include comparing a calculated value ( $\alpha$ ) to a threshold and determining the seal quality (i.e., *Padhi*'s score) between the two sides of the ear terminal ("comparing the value to a threshold to determine a seal quality between the first side and the second side").

**(vi) [1.2.7]: "generating a message to convey the seal quality; and sending the message to the user."**

177. In my opinion, *Epley* discloses or suggests "generating a message to convey the seal quality; and sending the message to the user."

178. As I explain above in connection with element [1.2], *Epley* discloses “instructions” “execut[ed] in a processor,” and those instructions control the operation of the feedback mechanism 24, including its seal test.

179. Moreover, *Epley* discloses a “display produces a warning to the operator” according to “whether the ear probe is properly sealed.” [0070].

Those of skill in the art will appreciate that microphone 76, amplifier 66 a/b and ADC 68 a/b that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ensure proper sealing of left and right ear probes. 18 a and 18 b. This is accomplished, for example, by applying a known sound and volume and measuring the acoustic response within the subject's ear as picked up by the microphone. A straightforward comparison of input and output sound parameters ***determine whether the ear probe is properly sealed*** and further testing is avoided if the seal is not secure. ***The display produces a warning to the operator.*** Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

*Epley*, [0070]. *Epley* further discloses that the display screen 36 enables “display[ing] and presentation of data, text and graphic representation.” *Epley*, [0024].

System 10 further includes a keyboard 34 including an associated cursor control device such as a mouse (not shown) to enable inputs to

and control of expert subsystem 26 by the operator. System 10 further includes a ***display screen 36 to enable display and presentation of data, text and graphic representations*** of stimulus/response or other subject or test condition data to the operator. As indicated in somewhat simplified form in the split screen schematic illustration of FIG. 1, display screen 36 can assist the operator in expertly analyzing ocular and/or sway response to postural and/or sound and/or pressure stimuli. Finally, system 10 includes a destabilization device 38 for intentionally posturally destabilizing a subject standing at least partly thereon, as will be described further below. Those of skill in the art will appreciate that, during a subject test session, different sound and/or pressure test parameters and/or criteria can be used to probe the subject, and different postural conditions, can be controlled by the operator to evoke different ocular or postural sway responses from the subject, in order to detect and perhaps diagnose and even treat a subject determined to suffer a vestibular disorder.

Epley, [0024].

180. To the extent the Board finds “seal quality” is not binary in nature, it would have been obvious to implement Epley to provide a non-binary “warning” conveying seal quality for the reasons I explain in connection with element [1.2.6].

181. Moreover, to the extent the “processor is configured to execute the instructions to perform operations, the operations comprising” limitation is determined to be a means-plus-function limitation, *Epley*’s disclosure is the same

or equivalent to the algorithm disclosed in the '493 patent at step 420, which requires “If the instantaneous cross-correlation of the two signal buffers is less than the threshold value 414, then the seal test status is set to FAIL 418 (i.e. an ear-seal leak is detected); otherwise, if the cross-correlation is suitably high, the seal test status is set to PASS 420.” ’493 patent, 9:48-52.

182. Accordingly, in my opinion, a person having ordinary skill in the art would have understood that *Epley* discloses or suggests that instructions executed by the processor and stored in the memory include generating and sending a message to the user that conveys seal quality to the user (“generating a message to convey the seal quality” and “sending the message to the user”).

**7. Claim 3: “The audio device according to claim 1, wherein the message is a visual message sent to a display of the audio device.”**

183. In my opinion, *Epley* in view of *Padhi* and *Voix* discloses or suggests “[t]he audio device according to claim 1, wherein the message is a visual message sent to a display of the audio device.”

184. As I explain above in connection with claim 1, *Epley* in view of *Padhi* and *Voix* disclose the “audio device according to claim 1,” including a message. See [1.2.7].

185. Moreover, *Epley* discloses a “display screen 36 to enable display and presentation of data, text and graphic representations.” [0024].

System 10 further includes a keyboard 34 including an associated cursor control device such as a mouse (not shown) to enable inputs to and control of expert subsystem 26 by the operator. System 10 further includes **a display screen 36 to enable display and presentation of data, text and graphic representations** of stimulus/response or other subject or **test condition data to the operator**. As indicated in somewhat simplified form in the split screen schematic illustration of FIG. 1, **display screen 36 can assist the operator** in expertly analyzing ocular and/or sway response to postural and/or sound and/or pressure stimuli. Finally, system 10 includes a destabilization device 38 for intentionally posturally destabilizing a subject standing at least partly thereon, as will be described further below. Those of skill in the art will appreciate that, during a subject test session, different sound and/or pressure test parameters and/or criteria can be used to probe the subject, and different postural conditions, can be controlled by the operator to evoke different ocular or postural sway responses from the subject, in order to detect and perhaps diagnose and even treat a subject determined to suffer a vestibular disorder.

*Epley*, [0024]. *Epley* further discloses that the “processed data can be presented in representational form on a display screen integral with the system.” *Epley*, [0060].

Those of skill in the art will appreciate that standardized and semi-automated testing and recording of clinical or field trials of subject stimuli/response represents an important advantage over prior art, ad hoc testing and reporting. Thus, system 10 facilitates subject vestibular condition testing and vestibular disorder detection,

diagnosis and treatment. Raw or processed data can be recorded for later oversight and analysis. More important, perhaps, ***processed data can be presented in representational form on a display screen integral with the system*** for instant analysis and detection by the operator or the machine (e.g. the computer), and for possible treatment. With the expert subsystem operating within computer 40, an operator can stimulate a subject in a measured and predetermined way and can measure and evaluate the subject's nystagmic and/or sway response thereto also in a measured and predetermined way. Thus, testing, detection and treatment are semi-automated and greatly facilitated. A permanent record can be made for archival purposes.

*Epley*, [0060]. Figure 2 illustrates display 36, which displays a visual message.

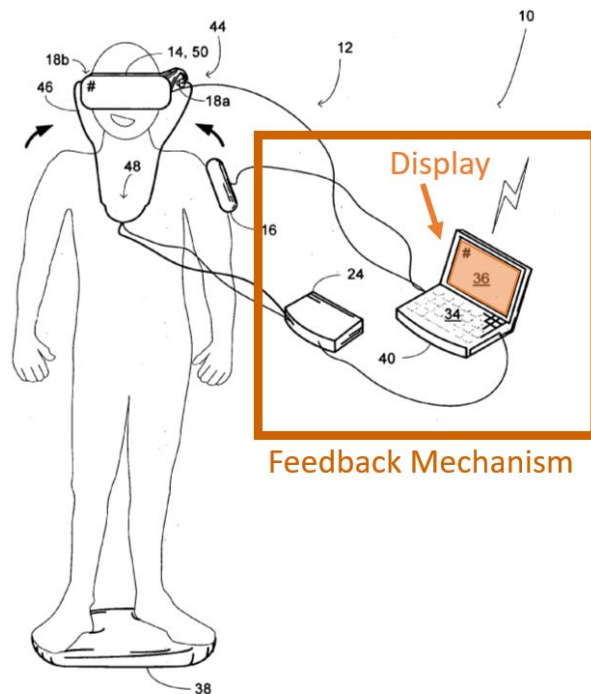


Fig. 2

Epley, Fig. 2

186. While mechanism 24 is illustrated as separate from computer 40 in, *Epley* explains that the “sound and pressure generator 24...can be separate from computer 40 or integrated therein.” *Epley*, [0029]. When integrated, the computer with the display 36 includes the functionality of the sound and pressure generator 24.

Those of skill in the art will appreciate that analyzers 20 and 22, expert subsystem 26, keyboard 34 and display 36 within the spirit and scope of the invention can take the form of a desktop or portable laptop computer 40, a personal digital assistant (PDA) or other small, lightweight, portable electronic device having at least data processing capability and preferably also having data displaying capability. Those of skill in the art also will appreciate that body-worn apparatus 12 includes a small, lightweight, portable headset 44 that at least partly integrates cameras 14 and ear probes 18 and one or more head/body sensors 16, as shown in FIG. 2. Indeed, either or both of the headset and sensor(s) can be battery operated. Those of skill in the art also will appreciate that ***sound and pressure generator 24 can be battery operated and can be separate from computer 40 or integrated therein***, e.g. in the form of a custom input/output (I/O) card that plugs into a motherboard or I/O slot. Moreover, those of skill in the art will appreciate that destabilization device 38 broadly refers to any device

particularly configured posturally to decrease the baseline stability of (i.e. to destabilize) a subject, e.g. it may take the form of a small step or incline or deformable footpad or cushion (e.g. a loosely fluid- or air-filled bladder, as shown) that, due at least in part to the subject's decreased baseline stability (e.g. balance shift) thereon tends to dis-equilibrate the subject. Thus, preferably device 38 is also small, lightweight and portable.

*Epley*, [0029].

187. *Epley* discloses that the display 36 “produces a warning to the operator,” which would be understood to be a visual message. *Epley*, [0070].

Those of skill in the art will appreciate that microphone 76, amplifier 66 a/b and ADC 68 a/b that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ensure proper sealing of left and right ear probes. 18 a and 18 b. This is accomplished, for example, by applying a known sound and volume and measuring the acoustic response within the subject's ear as picked up by the microphone. A straightforward comparison of input and output sound parameters determine whether the ear probe is properly sealed and further testing is avoided if the seal is not secure. ***The display produces a warning to the operator.*** Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

*Epley*, [0070].

188. Accordingly, in my opinion, a person having ordinary skill in the art would have understood that *Epley* discloses or suggests the audio device of claim 1, wherein a warning to the operator is presented on display 36 (“[t]he audio device according to claim 1, wherein the message is a visual message sent to a display of the audio device.”).

**8. Claim 4: “The audio device according to claim 1, wherein the speaker plays the test signal from the first side.”**

189. In my opinion, *Epley* in view of *Padhi* and *Voix* discloses or suggests “[t]he audio device according to claim 1, wherein the speaker plays the test signal from the first side.”

190. As I explain above in connection with claim 1, *Epley* in view of *Padhi* and *Voix* discloses the “audio device according to claim 1,” and in connection with elements [1.2.2] and [1.2.3], I explain inside the ear canal is the “first side.”

191. Moreover, *Epley* discloses “speakers 74” (one for each ear probe) that “deliver sound” “to the subject’s ears” (the first side). [0067]

Those of skill in the art will appreciate that the two-channel sound and pressure generators and ***speakers 74*** are used in accordance with the invention to ***deliver sound*** and/or pressure stimuli ***to the subject's ears***. They also can be used, within the spirit and scope of the invention, to deliver voice commands to the subject, in manual test mode, via live spoken words or phrases from an operator or, in automatic test mode, via digitized and recorded (and perhaps real-time

synthesized speech) words or phrases from the expert subsystem part of the computer. Those of skill in the art also will appreciate that, within the spirit and scope of the invention, microphone 76, amplifier 66 a/b and ADC 68 a/b can be omitted entirely from mechanism 24, while still permitting the sound and pressure generator portion of mechanism 24 to remain fully operational but without the benefit of feedback and monitoring. Such alternatives are contemplated as being within the spirit and scope of the invention.

[0067]. *Epley's* speakers 74 are secured within earpiece 70, which is fitted into the ear canal with an "ear-canal-easy cushion." [0065].

In accordance with one embodiment of the invention, air tube 72, ***speaker 74 and microphone 76 are secured within earpiece 70*** by potting compound 86, e.g. of epoxy resin. In accordance with one embodiment of the invention, potting compound 86 extends only partway toward the distal end of earpiece 70, thus forming hollow (air-filled) distal region 82 that is in fluid (air) communication with the subject's left ear. Alternatively, yet within the spirit and scope of the invention, the potting compound can be of any suitable material that instead may fill the entire interior region of earpiece 70. In accordance with one embodiment of the invention, earpiece 70 is of machined or extruded aluminum, although within the spirit and scope of the invention it can be of any suitable material. Those of skill in the art will appreciate that ***earpiece 70 is configured to have an ear-canal-easy cushion***, not shown, slipped over its distal end for the comfort of a test subject.

*Epley*, [0065]. Earpiece 70 forms “sealing engagement with the wall of the subject’s ear canal.” *Epley*, [0004]. Thus, the inside of the ear canal is the first side.

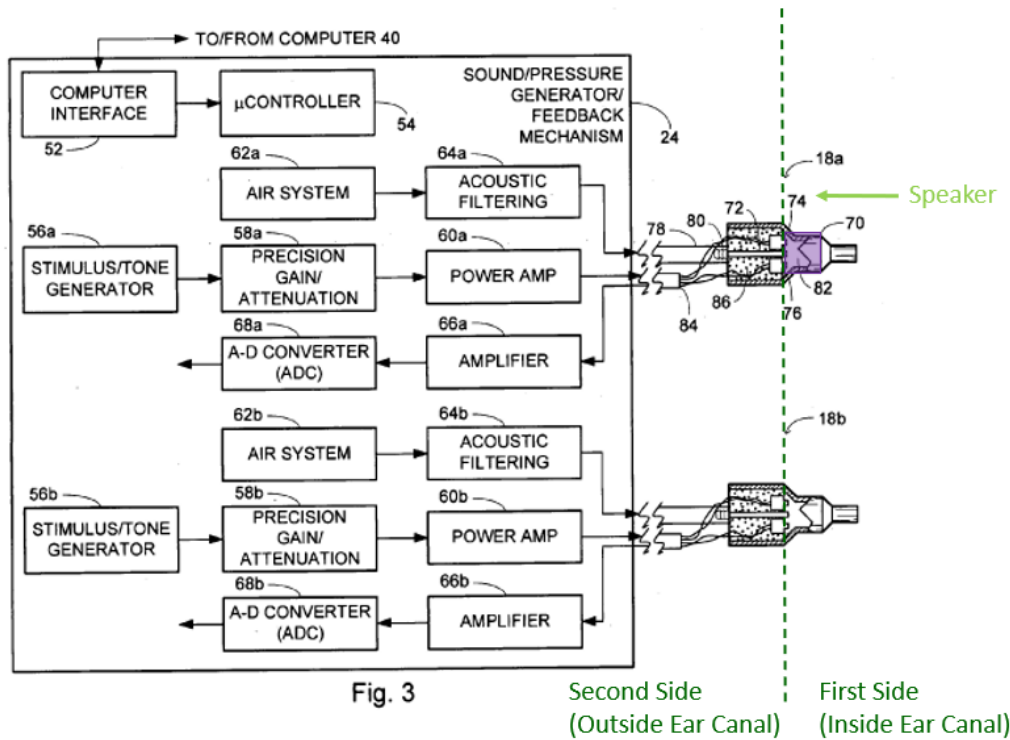
The invented apparatus for testing a subject's vestibular response to ear pressure or sound stimuli includes a headset including one or more ear-worn devices such a headphones or probes, wherein each probe is configured to be inserted into a subject's ear in an operational alignment with the subject's ear canal and ***sealing engagement with the wall of the subject's ear canal***, and a stimulus-producing mechanism operatively coupled with the headset configured to deliver defined pressure or sound stimuli to the one or more probes. The apparatus further includes one or both of 1) an ocular response analyzer operatively coupled to the headset, the ocular response analyzer measuring the subject's nystagmatic response to the stimuli by electronic, e.g. videographic, means and computerized comparison of potentially aberrational response data from the subject to recorded baseline subject data and 2) a sway response analyzer operatively coupled to the headset, the sway response analyzer including an inertial sensor of the subject's head and torso positional and orientational response to the stimuli and computerized comparison of potentially aberrational inertial data from the subject to recorded baseline inertial subject data. An optional destabilizing device for producing imbalance in a nominally stable subject thereby to test the subject's vestibular function and a carrying case configured and dimensioned portably to contain and transport the headset including

the probes, the stimulus-producing mechanism, the ocular response analyzer and the destabilizing device therein.

*Epley*, [0004]. Further, *Epley* discloses that the known sound (test signal) is applied within the subject's ear, the first side. [0070].

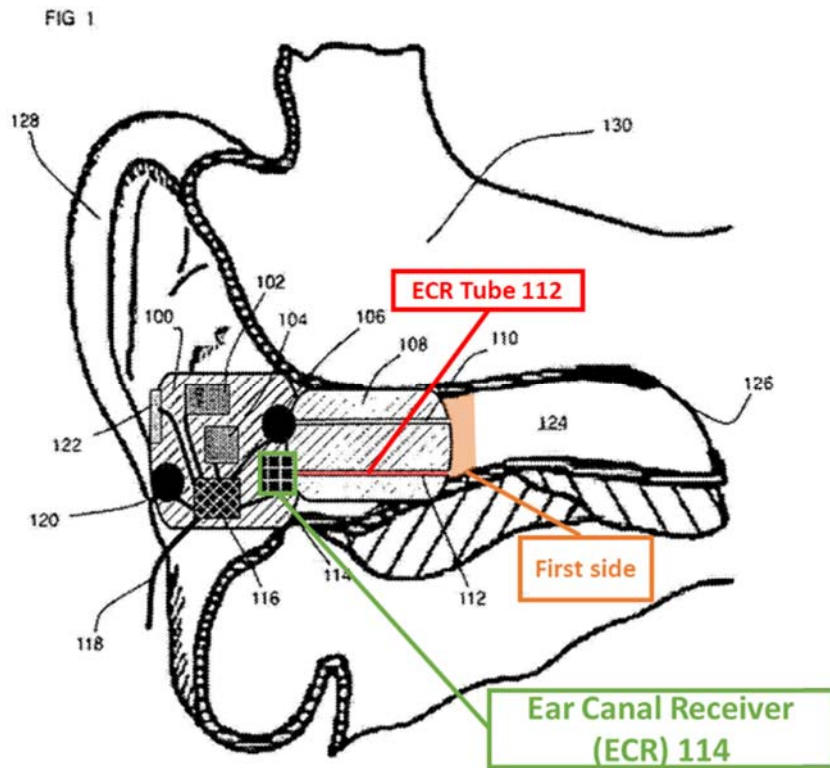
Those of skill in the art will appreciate that microphone 76, amplifier 66 a/b and ADC 68 a/b that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ensure proper sealing of left and right ear probes. 18 a and 18 b. This is accomplished, for example, by ***applying a known sound*** and volume and measuring the acoustic response ***within the subject's ear*** as picked up by the microphone. A straightforward comparison of input and output sound parameters determine whether the ear probe is properly sealed and further testing is avoided if the seal is not secure. The display produces a warning to the operator. Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

*Epley*, [0070]. Thus, a person of ordinary skill in the art would have understood that the inside of the ear canal is the first side. Figure 3 (as annotated) illustrates this.



*Epley*, Fig. 3. *Epley* explains that “The distal ends or so-called ‘snouts’ of speaker 74 and microphone 76 also extend to hollow distal region 82 of earpiece 70, as shown.” *Epley*, [0064].

192. In the same way, the ’493 patent discloses “ECR 114 outputs an acoustic signal through ECR tube 112 and into ear canal 124.” ’493 patent, 5:6-8.



'493 patent, Fig. 1 (annotated).

193. Accordingly, in my opinion, a person having ordinary skill in the art would have understood that *Epley* discloses or suggests the audio device of claim 1, wherein speaker 74 play the test signal from the inside of the ear canal (“[t]he audio device according to claim 1, wherein the speaker plays the test signal from the first side.”).

**9. Claim 5: “The audio device according to claim 4, wherein the first microphone measures sound from the first side.”**

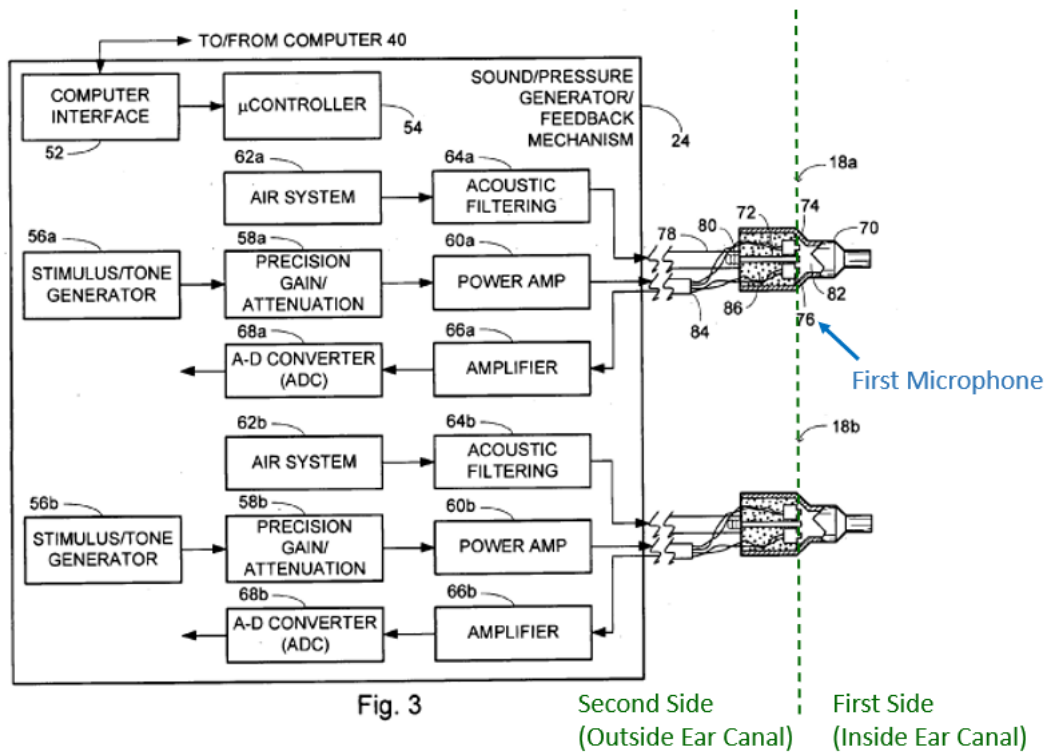
194. In my opinion, *Epley* in view of *Padhi* and *Voix* discloses or suggests “[t]he audio device according to claim 4, wherein the first microphone measures sound from the first side.”

195. As I explain above in connection with claim 1, *Epley* in view of *Padhi* and *Voix* disclose the “audio device according to claim 1,” and in connection with elements [1.2.2] and [1.2.3], I explain inside the ear canal is the “first side.”

196. Moreover, as I explained in [1.2.4], *Epley* discloses microphone 76 (the first microphone) measures sound “within the subject’s ear.” *Epley*, [0070].

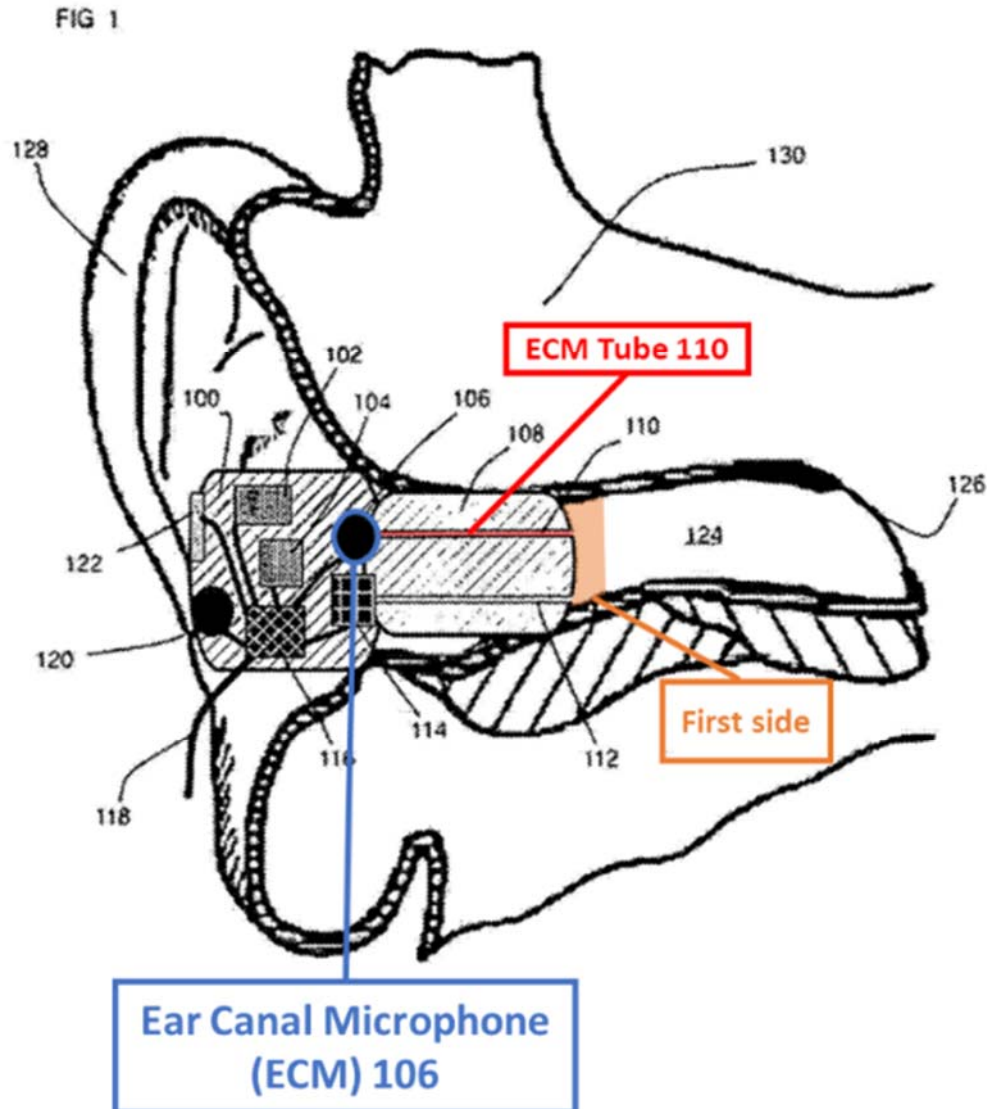
Those of skill in the art will appreciate that **microphone 76**, amplifier 66 a/b and ADC 68 a/b that form part of the feedback mechanism are used in accordance with one embodiment of the invention to ensure proper sealing of left and right ear probes. 18 a and 18 b. This is accomplished, for example, by applying a known sound and volume and **measuring the acoustic response within the subject's ear** as picked up by the microphone. A straightforward comparison of input and output sound parameters determine whether the ear probe is properly sealed and further testing is avoided if the seal is not secure. The display produces a warning to the operator. Thus a first use for this circuitry is during placement of the headset on the subject and during testing of the subject to ensure initial and continuously sealed acoustic engagement with the subject's ear canals.

Epley, [0070]. Figure 3 (as annotated) illustrates this.



Epley, Fig. 3. Epley explains that “The distal ends or so-called ‘snouts’ of speaker 74 and microphone 76 also extend to hollow distal region 82 of earpiece 70, as shown.” Epley, [0064].

197. In the same way, the ’493 patent discloses “ECM 106 receives an acoustic signal present in ear canal 124 though ECM tube 110.” ’493 patent, 5:9-11.



'493 patent, Fig. 1 (annotated).

198. Thus, a person of ordinary skill in the art would understand that microphone 76 measures acoustic response within the subject's ear canal.

199. Accordingly, in my opinion, a person having ordinary skill in the art would have understood that *Epley* discloses or suggests the audio device according to claim 4, wherein microphone 76 measures sound from within the ear canal

(“[t]he audio device according to claim 4, wherein the first microphone measures sound from the first side”).

**10. Claim 7: “The audio device according to claim 1, wherein the first signal is the test signal and the second signal is the first microphone signal.”**

200. In my opinion, *Epley* in view of *Padhi* and *Voix* disclose or suggest “[t]he audio device according to claim 1, wherein the first signal is the test signal and the second signal is the first microphone signal.”

201. The claimed “first signal” and “second signal” are relevant to element [1.2.5], which requires “***calculating a value, using a first signal and a second signal***, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination.”

202. As I explain above in connection with element [1.2.5] (§IX.A.6.d(iv)), *Epley* discloses a “comparison” of a “known sound” and an “acoustic response...picked up by the microphone” to “ensure proper sealing.”

203. While *Padhi* teaches various modifications to the input audio signals before the cross-correlation is performed, *Padhi*’s modifications are consistent with the requirement that the microphone and test signals themselves be used because the value calculated as a result of cross-correlation is ultimately obtained “using” the original audio signals. For example, the output of the cross-correlation function taught by *Padhi* will vary depending on changes to the “known sound”

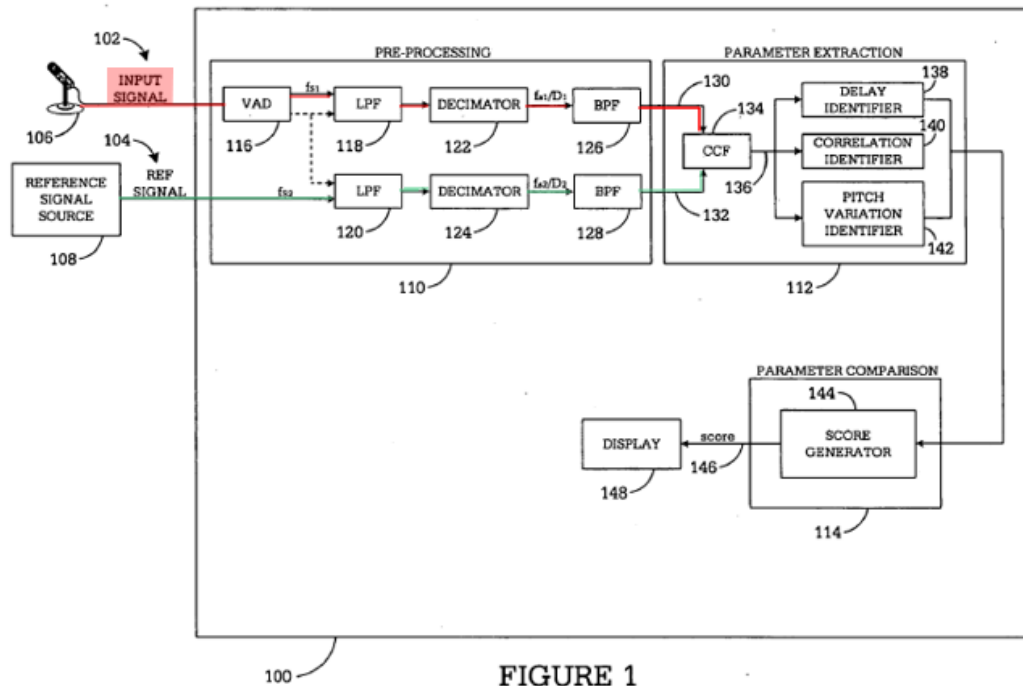
and “acoustic response...picked up by the microphone,” even if those signals pass through, *e.g.*, *Padhi*’s **pre-processing section 110**. For example, even though *Padhi* illustrates the cross-correlation function 134 is performed after passing through the **pre-processing section 110**, *Padhi* discloses using “the cross-correlated signal 136 to identify various characteristics associated with the input signal 102 and/or the reference signal 104.” *Padhi*, [0029]. A person having ordinary skill in the art would have understood that this is possible only because the cross-correlation function is performed using the input signal 102 (corresponding to *Epley*’s “acoustic response...picked up by the microphone”) and the reference signal 104 (corresponding to *Epley*’s “known sound”).

204. Regardless, to the extent the cross-correlation calculation must be directly performed on the test signal and first microphone signal without modification, it would have been a simple design choice to “omit[]” the pre-processing as it is not mandatory in *Padhi*. *See Padhi*, [0035]. Specifically, it is my opinion that inputting an audio signal into *Padhi*’s **cross correlation function 134** without first performing **pre-processing in section 110** (such that it is “the” signal and not a “modified” version) is a simple substitution of one known element (a “modified” pre-processed signal) for another known element (“the” signal without pre-processing) to yield predictable results—and vice versa.

205. Whether one, both, or neither signal is pre-processed in *Padhi*'s **section 110**, it is input into *Padhi*'s **cross correlation function 134**, and it is my opinion that a person having ordinary skill in the art would have understood that the result would be cross correlating two signals to obtain a **cross-correlated signal 136**. To the extent the cross correlation calculation must be directly performed on the test signal and first microphone signal without modification, it would have been a simple design choice to "omit[]" the pre-processing as it is not mandatory in *Padhi*. *Padhi*, [0035].

Although FIG. 1 illustrates one example of an audio apparatus 100, various changes may be made to FIG. 1. For example, the functional division of the apparatus 100 is for illustration only. ***Various components can be omitted or combined and additional components can be added according to particular needs.*** As particular examples, the voice activity detector 116 could be omitted, and the reference signal source 108 could form part of the apparatus 100. Also, the ***filters 118, 126*** may be combined, and the ***filters 120, 128*** may be combined. Further, the input signal 102 and reference signal 104 could represent analog or digital signals. If one or both signals 102, 104 are analog signals, one or more amplifiers and/or one or more analog-to-digital converters could be used to convert the signals 102, 104 into a digital format. In addition, the apparatus 100 could form part of a larger apparatus or system, such as a karaoke system or a computer system.

*Padhi*, [0035].



*Padhi*, Fig. 1

Thus, it is my opinion that there are no technical reasons why pre-processing must be performed, and making the substitution would have been routine and well within the engineering skill level of a person having ordinary skill in the art. Moreover, *Padhi* supports my opinion, as *Padhi* discloses that the components in pre-processing [section 110](#) can be omitted. *Padhi*, [0035].

206. Thus, a person of ordinary skill in the art would have understood that modification of the signals could be omitted or added according to particular needs. Moreover, a person of ordinary skill in the art would have understood that the advantages and tradeoffs of making the substitutions were predictable. For

example, omitting pre-processing would have the advantage of being more efficient/less computationally intensive. Further, a person of ordinary skill in the art would have understood that any such tradeoff would result in a straightforward design choice.

207. For example, *Padhi* describes the advantages of modifying a signal, including anti-aliasing of the signal, which would result in artifact-reduction. *Padhi*, [0023]-[0025].

In the pre-processing section 110, ***the input signal 102 is supplied to a filter 118, and the reference signal 104 is supplied to a filter 120.*** The filters 118, 120 filter the signals 102, 104 before the signals 102, 104 are processed further. Each filter 118, 120 includes any hardware, software, firmware, or combination thereof for filtering a signal. As an example, the filters 118, 120 may represent anti-aliasing low pass filters. As a particular example, ***the filters 118, 120*** may represent ***anti-aliasing low pass filters*** having a cut-off frequency of 3 kHz. In some embodiments, the filters 118, 120 do not process the signals 102, 104 until the voice activity detector 116 detects a voice in the input signal 102. The filter 118 is coupled to a decimator 122, and the filter 120 is coupled to a decimator 124. ***The decimators 122, 124 decimate the input signal 102 and the reference signal 104, which lowers the sampling frequency of the signals 102, 104.*** For example, the ***decimator 122 may decimate the input signal 102*** by selecting and outputting every other sample in the input signal 102. In some embodiments, the decimator 122 decimates the input signal 102 by a

first factor D1, and the decimator 124 decimates the reference signal 104 by a second factor D2. In these embodiments, if the input signal 102 has a frequency of  $f_{S1}$ , the decimated input signal 102 has a frequency of  $f_{S1}/D1$ . Similarly, if the reference signal 104 has a frequency of  $f_{S2}$ , the decimated reference signal 104 has a frequency of  $f_{S2}/D2$ . ***The decimators 122, 124 may decimate the signals 102, 104 by any suitable factors.*** In particular embodiments, the decimators 122, 124 decimate the signals 102, 104 so that the decimated signals 102, 104 have a frequency of 6 kHz. As an example, the input signal 102 and reference signal 104 may have sampling rates of 12,000 samples per second and 48,000 samples per second, respectively. If the decimated signals should have a frequency of 6 kHz, the decimator 122 has a decimation factor D1 of two, and the decimator 124 has a decimation factor D2 of eight. Each decimator 122, 124 includes any hardware, software, firmware, or combination thereof for decimating a signal 102, 104. A filter 126 is coupled to the decimator 122, and a filter 128 is coupled to the decimator 124. ***The filters 126, 128 filter the decimated input signal 102 and decimated reference signal 104. The filters 126, 128 filter the decimated signals 102, 104 before the decimated signals 102, 104 are processed further. Each filter 126, 128 includes any hardware, software, firmware, or combination thereof for filtering a signal. As an example, the filters 126, 128 may represent band pass filters. As a particular example, the filters 126, 128 may represent infinite impulse response (IIR) filters that band limit the decimated signals***

***102, 104 to 1 kHz, which helps to attenuate frequencies above 1.5 kHz.***

*Padhi*, [0023]-[0025].

208. Further, a person of ordinary skill in the art would have understood that, while there are benefits to filtering and decimating the microphone signal because that signal is unpredictable, it would have been a simple design choice to generate the test signal (which would be a predictable signal) such that filtering and decimation are not required, and thus cross-correlation is performed directly on the unmodified test signal.

209. Thus, as to each audio signal, a person having ordinary skill in the art would have two predictable choices: (1) omit the pre-processing as *Padhi* suggests may be done; or (2) include **pre-processing section 110**. It is my opinion that a person having ordinary skill in the art would have been motivated to make either of the design choices to achieve the claimed limitations. For example, by not including the additional circuitry from **pre-processing section 110**, a person having ordinary skill in the art would have recognized that such a system would have a simpler design with fewer computational requirements.

210. Further, it is my opinion that a person having ordinary skill in the art would have had a reasonable expectation of success in making these predictable substitutions/choices. Indeed, as I noted above, it is my opinion that a person

having ordinary skill in the art would have recognized that, regardless of whether one, both, or neither signal is pre-processed in *Padhi*'s [section 110](#), it is input into *Padhi*'s **cross correlation function 134**, and the result would be cross correlating two signals to obtain a **cross-correlated signal 136**.

211. Accordingly, in my opinion, a person having ordinary skill in the art would have understood that *Epley* in view of *Padhi* discloses or suggests the audio device according to claim 1 (see, claim 1), wherein the first signal is *Epley*'s “known sound” and the second signal is *Epley*'s “acoustic response picked up by the microphone” (“wherein the first signal is the test signal and the second signal is the first microphone signal”).

**11. Claim 8: “The audio device of claim 1, wherein the first signal is the test signal and the second signal is a modified first microphone signal.”**

212. In my opinion, *Epley* in view of *Padhi* and *Voix* disclose or suggest “[t]he audio device of claim 1, wherein the first signal is the test signal and the second signal is a modified first microphone signal.”

213. The claims “first signal” and “second signal” are relevant to the element [1.2.5], which requires “***calculating a value, using a first signal and a second signal***, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination.”

214. As I explain above in connection with element [1.2.5] (§IX.A.6.d(iv)), Epley discloses a “comparison” of a “known sound” and an “acoustic response...picked up by the microphone” to “ensure proper sealing.”

215. In addition, *Padhi* discloses, when comparing a microphone signal with a reference signal, the microphone signal is modified by “filter 118,” “filter 126” and “decimator 122.” *See Epley*, Fig. 1.

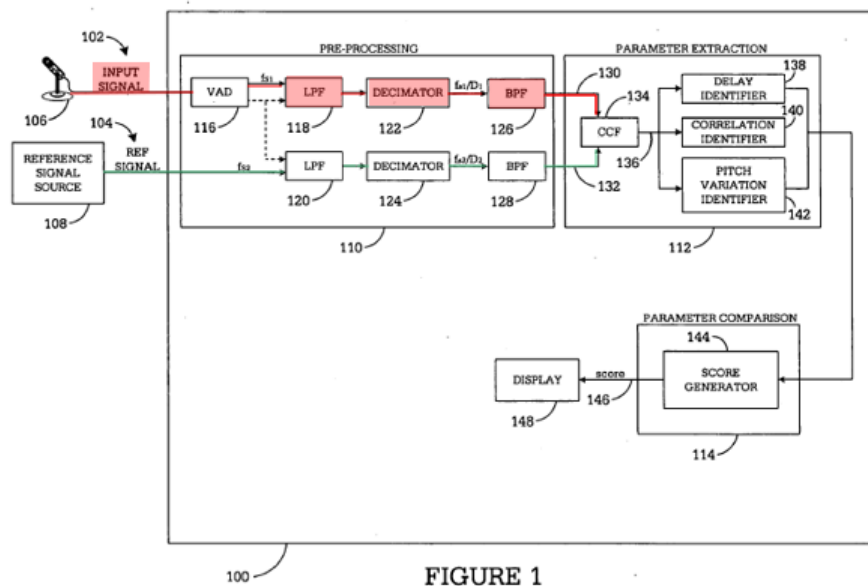
216. As discussed in (§IX.A.3), and [7] (§IX.A.10), a person having ordinary skill in the art would have been motivated to apply *Padhi*’s teachings of comparing audio signals through cross-correlation to *Epley*’s seal analyzer. Moreover, *Padhi* describes the advantages of modifying a signal, including anti-aliasing of the signal, which would result in noise-reduction. *See, e.g.*, *Padhi*, [0023]-[0025].

In the pre-processing section 110, ***the input signal 102 is supplied to a filter 118, and the reference signal 104 is supplied to a filter 120.*** The filters 118, 120 filter the signals 102, 104 before the signals 102, 104 are processed further. Each filter 118, 120 includes any hardware, software, firmware, or combination thereof for filtering a signal. As an example, the filters 118, 120 may represent anti-aliasing low pass filters. As a particular example, ***the filters 118, 120*** may represent ***anti-aliasing low pass filters*** having a cut-off frequency of 3 kHz. In some embodiments, the filters 118, 120 do not process the signals 102, 104 until the voice activity detector 116 detects a voice in the

input signal 102. The filter 118 is coupled to a decimator 122, and the filter 120 is coupled to a decimator 124. ***The decimators 122, 124 decimate the input signal 102 and the reference signal 104, which lowers the sampling frequency of the signals 102, 104.*** For example, the ***decimator 122 may decimate the input signal 102*** by selecting and outputting every other sample in the input signal 102. In some embodiments, the decimator 122 decimates the input signal 102 by a first factor D1, and the decimator 124 decimates the reference signal 104 by a second factor D2. In these embodiments, if the input signal 102 has a frequency of  $f_{S1}$ , the decimated input signal 102 has a frequency of  $f_{S1}/D1$ . Similarly, if the reference signal 104 has a frequency of  $f_{S2}$ , the decimated reference signal 104 has a frequency of  $f_{S2}/D2$ . ***The decimators 122, 124 may decimate the signals 102, 104 by any suitable factors.*** In particular embodiments, the decimators 122, 124 decimate the signals 102, 104 so that the decimated signals 102, 104 have a frequency of 6 kHz. As an example, the input signal 102 and reference signal 104 may have sampling rates of 12,000 samples per second and 48,000 samples per second, respectively. If the decimated signals should have a frequency of 6 kHz, the decimator 122 has a decimation factor D1 of two, and the decimator 124 has a decimation factor D2 of eight. Each decimator 122, 124 includes any hardware, software, firmware, or combination thereof for decimating a signal 102, 104. A filter 126 is coupled to the decimator 122, and a filter 128 is coupled to the decimator 124. ***The filters 126, 128 filter the decimated input signal 102 and decimated reference signal 104. The filters 126, 128 filter***

*the decimated signals 102, 104 before the decimated signals 102, 104 are processed further. Each filter 126, 128 includes any hardware, software, firmware, or combination thereof for filtering a signal. As an example, the filters 126, 128 may represent band pass filters. As a particular example, the filters 126, 128 may represent infinite impulse response (IIR) filters that band limit the decimated signals 102, 104 to 1 kHz, which helps to attenuate frequencies above 1.5 kHz.*

Padhi, [0023]-[0025]



Padhi, Fig. 1

217. As discussed in [7], to the extent the cross-correlation calculation must be directly performed on the test signal without modification, it would have been a simple design choice to “omit[]” the pre-processing as it is not mandatory in

*Padhi*. See §IX.A.10. Moreover, to the extent this claim requires that the cross-correlation calculation be performed directly on the test signal without modification, as I explain in connection with claim [7], a person having ordinary skill in the art would have understood, while there are benefits to filtering and decimating the microphone signal because that signal is unpredictable, it would have been a simple design choice to generate the test signal (which would be a predictable signal) such that filtering and decimation are not required, and thus cross-correlation is performed directly on the unmodified test signal.

218. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Epley* in view of *Padhi* disclose or suggest the audio device of claim 1 (*see*, claim 1), wherein the first signal is *Epley*'s "known sound" and the second signal is *Epley*'s "acoustic response picked up by the microphone" as modified by filtering and decimating as described by *Padhi* ("wherein the first signal is the test signal and the second signal is a modified first microphone signal").

**12. Claim 10: "The audio device of claim 1, wherein the first signal is a modified test signal and the second signal is the first microphone signal."**

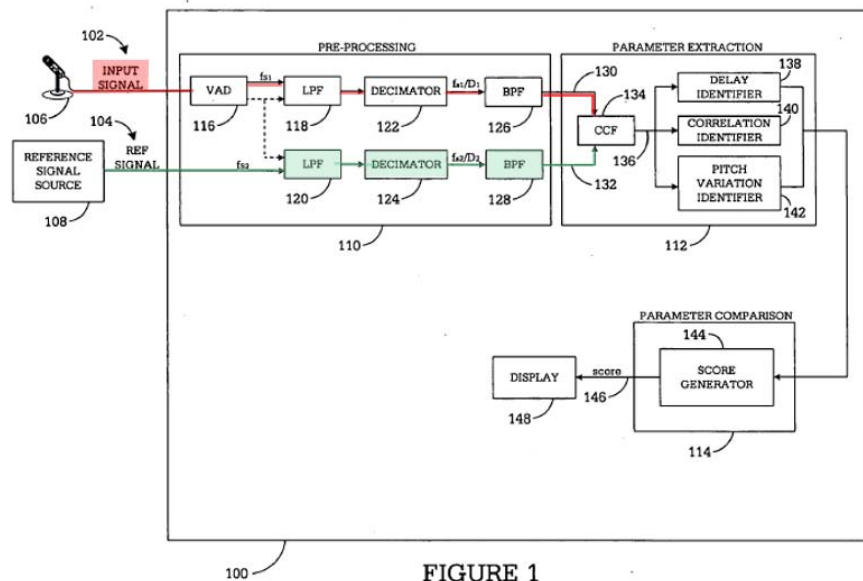
219. In my opinion, *Epley* in view of *Padhi* and *Voix* disclose or suggest "[t]he audio device of claim 1, wherein the first signal is a modified test signal and the second signal is the first microphone signal."

220. The claims “first signal” and “second signal” are relevant to the element [1.2.5], which requires “***calculating a value, using a first signal and a second signal***, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination.”

221. As I explain above in connection with element [1.2.5] (§IX.A.6.d(iv)), Epley discloses a “comparison” of a “known sound” and an “acoustic response...picked up by the microphone” to “ensure proper sealing.”

222. Moreover, *Padhi* discloses, when comparing a microphone signal with a reference signal, the microphone signal is modified by “filter 120,” “filter 128” and “decimator 124.” *See, e.g.*, *Padhi*, [0023]-[0025].

223. As I explain in connection with claim 7, to the extent the cross-correlation calculation must be directly performed on the first microphone signal without modification, it would have been a simple design choice to “omit[]” the pre-processing, which is not mandatory in *Padhi*. *See* §IX.A.10; *Padhi*, Fig. 1.



*Padhi*, Fig. 1

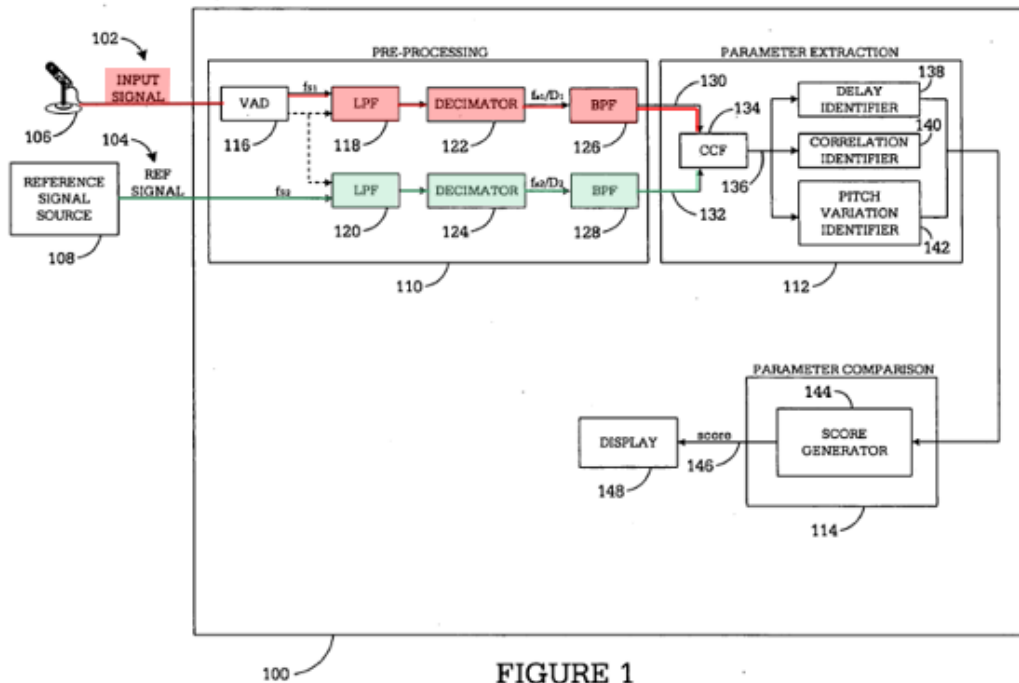
224. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Epley* in view of *Padhi* disclose or suggest the audio device of claim 1 (see, claim 1), wherein the first signal is *Epley*’s “known sound” as modified by filtering and decimating as described by *Padhi* and the second signal is *Epley*’s “acoustic response picked up by the microphone” (“wherein the first signal is a modified test signal and the second signal is the first microphone signal”).

**13. Claim 11: “The audio device according to claim 1, wherein the first signal is a modified test signal and the second signal is a modified first microphone signal.”**

225. In my opinion, *Epley* in view of *Padhi* and *Voix* disclose or suggest “[t]he audio device according to claim 1, wherein the first signal is a modified test signal and the second signal is a modified first microphone signal.”

226. The claims “first signal” and “second signal” are relevant to the element [1.2.5], which requires “***calculating a value, using a first signal and a second signal***, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination.”

227. As I explain above in connection with claims 8 and 10, it would have been obvious to supply *Epley*’s known sound and acoustic response to *Padhi*’s **pre-processing section 110**—creating a “modified” version of them with, among other operations, a low pass filter—and supplying “modified” signals to *Padhi*’s **cross correlation function 134**, and it would have been obvious to use these signals**Error! Reference source not found..** See §§IX.A.11, IX.A.12; *Padhi*, Fig. 1.



*Padhi*, Fig. 1

228. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Epley* in view of *Padhi* disclose or suggest the audio device of claim 1 (see, claim 1), wherein the first signal is *Epley*’s “known sound” as modified by filtering and decimating as described by *Padhi* and the second signal is *Epley*’s “acoustic response picked up by the microphone” as modified by filtering and decimating as described by *Padhi* (“wherein the first signal is a modified test signal and the second signal is a modified first microphone signal”).

**14. Claim 18: “The audio device of claim 1, wherein the seal quality is good if the value is at or above the threshold value.”**

229. In my opinion, *Epley* in view of *Padhi* and *Voix* discloses or suggests “[t]he audio device of claim 1, wherein the seal quality is good if the value is at or above the threshold value.”

230. As I explain above in connection with claim 1, *Epley* in view of *Padhi* and *Voix* disclose claim 1.

231. Further, *Voix* discloses that the “presence of acoustic seal” occurs when the attenuation value is “equal to or larger than a predetermined threshold value.” [0039].

Typically, step d) includes indicating ***presence of acoustic seal when attenuation between said first and second sound levels is equal to or larger than a predetermined threshold value*** within a predetermined frequency range.

*Voix*, [0039].

232. When the attenuation is high, this indicates that the sound outside the ear canal is very different than the sound inside the ear canal, indicating a strong acoustic seal. *Voix* describes this as “a good acoustic seal.” *Voix*, [0147] (“The above detailed acoustic seal test ensures that the in-ear device 12 or earplug fits

properly inside the ear canal 14 of the user, thereby providing *a good acoustic seal.*”)

233. This is consistent with the application of *Padhi*’s cross-correlation to *Epley*’s comparison of a test signal and the feedback from the microphone inside the ear canal. Cross-correlation is a measure of the similarity of two signals, and a person of ordinary skill in the art would have understood that higher cross-correlation occurs when the two signals are nearly identical, and therefore the test signal and microphone signal will be nearly identical. This similarity between the test signal and the signal picked up by the microphone occurs when the sound inside the ear and picked up by the microphone inside the ear canal is not significantly impacted by additional noise that enters the ear canal, that is, when there is a good acoustic seal. *Voix*, [0147]. Likewise, a person of ordinary skill in the art would have understood that lower cross-correlation occurs when the two signals are less similar, and therefore the test signal and microphone signal will be less similar when there is a bad acoustic seal because more external sound enters the ear canal if the acoustic seal is bad.

234. As I explain above in Section IX.A.5, a person of ordinary skill in the art would have been motivated to apply *Voix*’s known teachings of a threshold value in implementing *Epley*’s feedback mechanism.

235. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Epley* in view of *Padhi* and *Voix* to disclose or suggest the audio device of claim 1 (see claim 1), wherein a good acoustic seal is present if the value is at or above the threshold value (“wherein the seal quality is good if the value is at or above the threshold value.”).

**15. Claim 19: “The audio device of claim 1, wherein the sealing section is composed of at least one of rubber, silicone, a flexible polymer, or a combination thereof.”**

236. In my opinion, *Epley* in view of *Padhi* and *Voix* discloses or suggests “[t]he audio device of claim 1, wherein the sealing section is composed of at least one of rubber, silicone, a flexible polymer, or a combination thereof.”

237. *Voix* discloses the in-ear device is “made out of” “silicones” because it is a “soft material[.]” [0099].

Referring now in more detail to FIG. 1, there is shown an apparatus 10 for assessing in situ acoustical performance of an in-ear device 12 in accordance with an embodiment of the present invention. ***The in-ear device 12 or earplug, typically made out of generally soft materials such as silicones,*** acrylic materials and the like, is typically first inserted inside the ear canal 14 of the ear 16 of an individual (user or wearer) and then fitted to the contour thereof using a thermosettable compound 18 (shown in dotted lines) injected inside the in-ear device 12. The earplug 12 typically includes at least one

passageway 20 or sound bore that extends there through to be in fluid communication with the ear canal 14 when the earplug 12 is inserted therein. Although an earplug 12 is being shown throughout the Figures for illustration purposes, one skilled in the art would understand that any type of in-ear device (HPDs, HADs or the like) could be used without departing from the scope of the present invention.

Voix, [0099].

238. As discussed in §IX.A.5, a person of ordinary skill in the art would have been motivated to apply *Voix*'s known teachings of making in-ear devices “out of soft materials such as silicones” in implementing *Epley*'s earpiece.

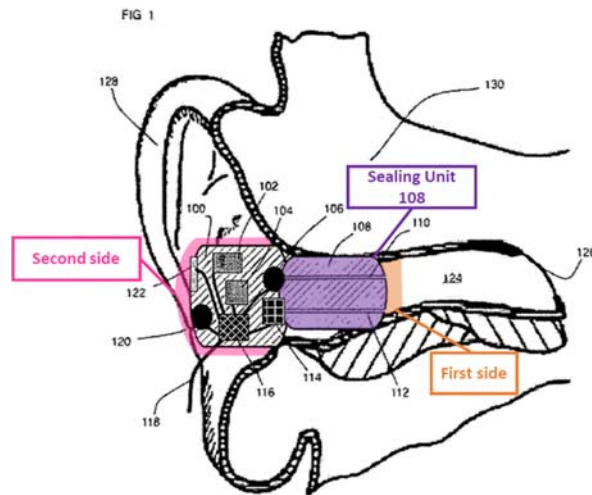
239. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Epley* in view of *Padhi* and *Voix* to disclose or suggest the audio device of claim 1 (see claim 1), wherein the device is made out of silicone (“wherein the sealing section is composed of at least one of rubber, silicone, a flexible polymer, or a combination thereof”).

**16. Claim 20: “The audio device of claim 1, wherein the sealing section is an eartip.”**

240. In my opinion, *Epley* in view of *Padhi* and *Voix* to discloses or suggests “[t]he audio device of claim 1, wherein the sealing section is an eartip.”

241. The '493 patent's specification does not use the term “eartip.” In my opinion, and as shown by Figure 1 of the '493 patent, the '493 patent discloses an

ear tip that is a component that seals the **first side** of the ear terminal from the **second side**.



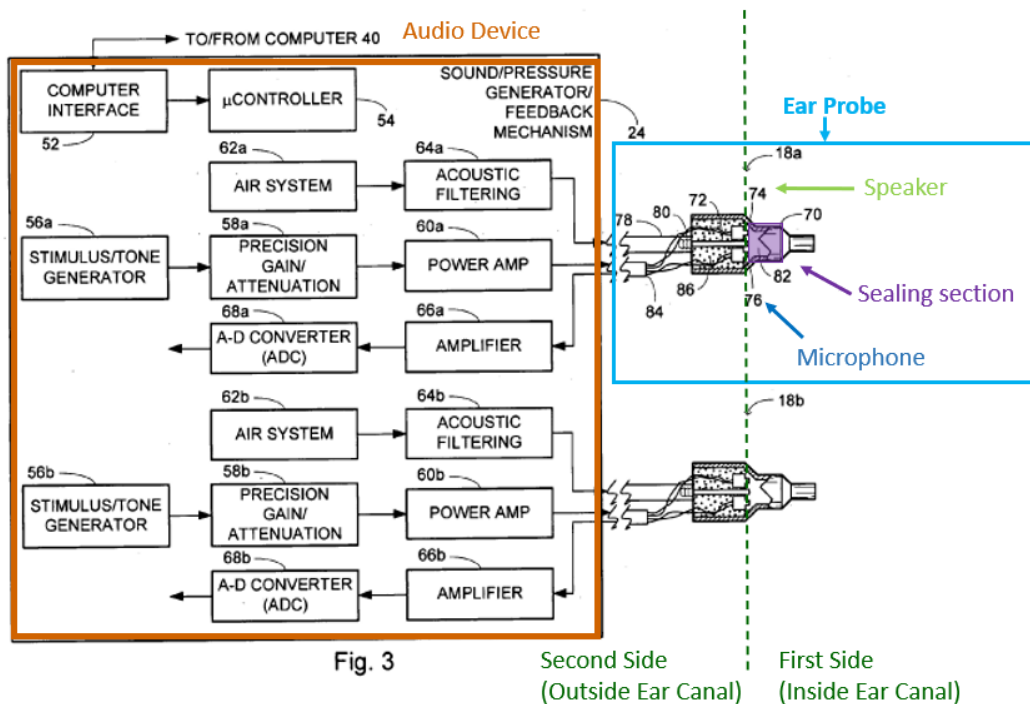
'493 patent, Fig. 1.

242. *Epley* discloses that “earpiece 70 is configured to have an ear-canal-easy cushion” that is “slipped over its distal end for the comfort of a test subject.” See [1.2.3].

In accordance with one embodiment of the invention, air tube 72, speaker 74 and microphone 76 are secured within earpiece 70 by potting compound 86, e.g. of epoxy resin. In accordance with one embodiment of the invention, potting compound 86 extends only partway toward the distal end of earpiece 70, thus forming hollow (air-filled) distal region 82 that is in fluid (air) communication with the subject's left ear. Alternatively, yet within the spirit and scope of the invention, the potting compound can be of any suitable material that instead may fill the entire interior region of earpiece 70. In

accordance with one embodiment of the invention, earpiece 70 is of machined or extruded aluminum, although within the spirit and scope of the invention it can be of any suitable material. Those of skill in the art will appreciate that ***earpiece 70 is configured to have an ear-canal-easy cushion, not shown, slipped over its distal end for the comfort of a test subject.***

*Epley*, [0065]. The tapered shape of earpiece 70 is illustrated in Figure 3, as annotated below.

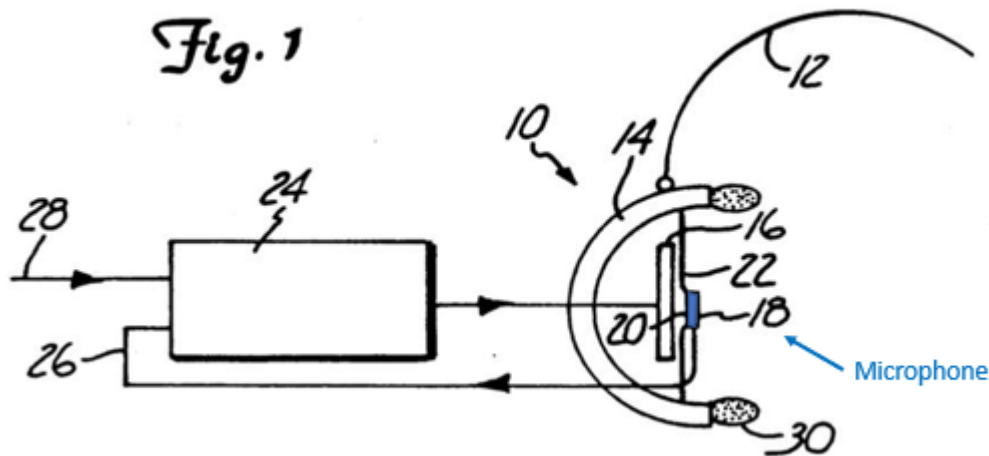


243. Thus, in my opinion, a person having ordinary skill in the art would have understood that *Epley* discloses that the sealing section is an eartip in the same way as the '493 patent.

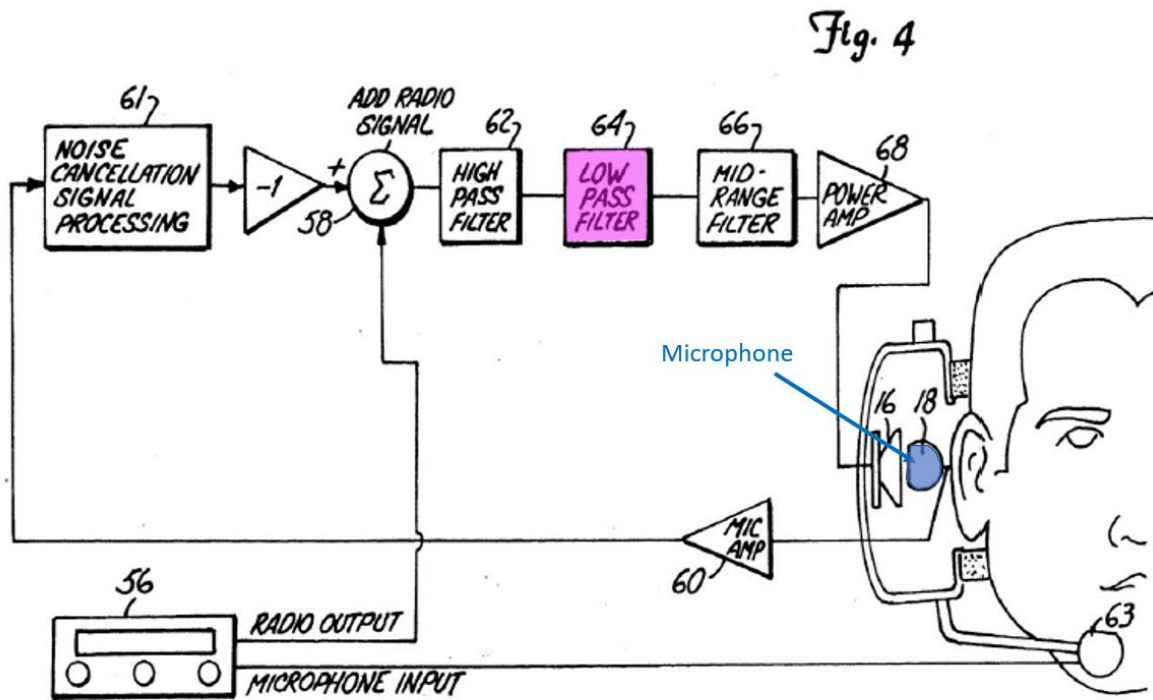
244. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Epley* in view of *Padhi* and *Voix* disclose or suggest earpiece 70 with ear-canal-easy cushion, slipped over its distal end (“sealing section is an eartip”).

**B. Ground 2: *Epley*, *Padhi*, *Voix*, and *Bourk* Render Obvious Claims 9 and 12**

245. *Bourk* is titled “noise cancellation headset.” Figures 1 and 4 illustrate headset 10, microphone 18, and low pass filter 64. *Bourk*, 3:50-4:27; *id.*, 6:37-7:17.



*Bourk*, Fig. 1.



*Bourk, Fig. 4.*

As will be more completely described with regard to the figures, the present invention is embodied in a new and novel headset system shown in FIG. 1 and generally designated 10. It will be noted that only the right side of the headset is shown and described, since the right and left side are mirror images of each other.

Headset 10 is shown to include headband 12, earcup 14 which is mounted in any known manner to headband 12, driver 16 mounted within earcup 14 which receives and acoustically reproduces desired electronic signals, and **microphone 18** having a vented face 20. **Microphone 18** generates a signal representative of the acoustic reproductions from driver 16 and background or ambient noise which enters the cavity formed between earcup 14 and the

wearer. **Microphone 18** is attached to grille plate 22 by any appropriate means so that microphone 18 is centrally positioned over driver 16 and wherein vented face 20 is oriented to point towards driver 16. As will become apparent in relation to FIGS. 2 and 3, **microphone 18** is positioned in relation to driver 16 so that it is acoustically closely coupled thereto. Grille plate 22 is attached to earcup 14 and extends across the front face of driver 16. In the preferred embodiment, microphone 18 is a standard electret microphone such as a Knowles 1759.

As indicated above, the signal generated by **microphone 18** is provided to electronic circuit 24, at input 26. Electronic circuit 24 processes the desired audio signal provided at input 28, and generates a modified signal for acoustical reproduction by the headset. The modified signal is generated by summing the original audio signal (if any) and an anti-noise signal that is generated based on the signal obtained from **microphone 18**. The modified signal is thereafter provided to driver 16. Since the anti-noise signal is added to the original audio signal, a portion of the modified signal reproduced by driver 16 will have the effect or tendency to cancel or null background noise in the earcup cavity; also, if the frequency response of the microphone is relatively linear the microphone may detect distortion caused by the speaker as noise and effectively reduce or cancel such distortion as well. As will be more particularly described in relation to FIG. 4, the modified signal may be passed through a plurality of cascaded filters for accentuating a desired frequency range prior to being provided to driver 16.

*Bourk, 3:50-4:27.*

As shown in FIG. 4, the series of cascaded filters may include a high pass filter 62 for filtering out those components of the modified audio signal having frequencies below a first frequency, and a **low pass filter 64 for filtering out those components of the modified audio signal having frequencies above a second frequency**. In the preferred embodiment the first frequency is in a range extending from 17 to 70 Hz and the ***second frequency is also in a range extending from 150 Hz to 8 kHz***. It should be understood that in filter 62 frequency components of the modified audio signal falling below 17 Hz will be attenuated by a fixed amount and that attenuation of the modified audio signal will decrease linearly for frequency components falling between 17 Hz to 70 Hz at which no attenuation takes place. Similarly, in **filter 64** frequency components of the modified audio signal falling above 8 kHz will be attenuated by a fixed amount and that attenuation of the modified audio signal increases linearly for frequency components falling between 150 Hz and 8 kHz at which maximum attenuation takes place. The 8 kHz value, a so-called pole, was chosen for the preferred embodiment to minimize phase lag around the frequency of 2.5 kHz.

The series of filters may also include a mid-range filter 66 for providing additional attenuation to those components of the modified audio signal having frequencies falling between a third and fourth frequency. In the preferred embodiment, the third frequency is 600 Hz and the fourth frequency is 1200 Hz. The centering of this range at

850 Hz provides a second order roll-off in a range where some phase lag can be tolerated.

The filtered modified audio signal is then provided to power amplifier 68 for amplification to a level appropriate for driver 16.

As a result of the above described invention, background or ambient noise reduction within the earcup as a function of frequency is predictable from the open loop frequency response of the headset system. ***Maximum noise reduction will occur at frequencies from about 30 Hz to about 200 Hz at values of approximately 15 to 28 dB.*** Above and below the 30 to 200 Hz frequency range, noise reduction falls off according to characteristics determined by the selection of filter parameters. Conventional passive noise reduction built into the headset, however, provides fairly effective noise reduction at higher frequencies, making the headset quite effective in achieving the desired overall reduction of noise, particularly for aircraft applications.

*Bourk*, 6:37-7:17.

246. As disclosed in the excerpts above, *Bourk* discloses generating a modified signal that is provided to driver 16 for acoustical reproduction by the headset. *Bourk* generates the modified signal by summing an audio signal (if any) and an anti-noise signal generated based on the signal from **microphone 18**. *Id.*, 4:7-16. *Bourk* discloses that “noise cancellation operates whether or not any such audio signal is provided,” such that the modified signal provided to driver 16 may

consist solely of the anti-noise signal generated from **microphone 18**. *Id.*, 5:45-49; *id.*, 6:12-17.

247. *Bourk* further discloses that the modified signal may be filtered before it is provided to driver 16. *Id.*, 4:23-27; *id.*, 6:25-35. The filters may include, e.g., a high pass filter 62 for filtering components below a first frequency and a **low pass filter 64 for filtering components above a second frequency**. *Id.*, 6:37-42. *Bourk* discloses that in the preferred embodiment, the first frequency is in the 17 to 70 Hz range and the second frequency is a range from 150 Hz to 8 kHz. *Id.*, 6:42-59.

248. *Bourk* further discloses that “[m]aximum noise reduction will occur at frequencies from about 30 Hz to about 200 Hz at values of approximately 15 to 28 dB.” *Id.*, 7:4-17. A person having ordinary skill in the art would have understood from this disclosure that *Bourk* suggests choosing an upper cutoff frequency of 200 Hz, that is, that the second frequency range filtered by the low pass filter 64 would be 200 Hz to 8 kHz.

### **1. Motivation to apply *Bourk* to *Epley, Padhi* and *Voix***

249. In my opinion, like *Epley, Padhi*, and *Voix*, *Bourk* is analogous art, is in the same field as the '493 patent, and is reasonably pertinent to alleged problems addressed by the '493 patent in processing audio signals in acoustic devices. For example, the '493 patent discloses it is directed to “audio processing circuitry.” '493 patent, Abstract; see also 4:59-62 (“Microprocessor 116 (or

processor 116) can be a logic circuit, a digital signal processor, controller, or the like for performing calculations and operations for the earpiece.”); *Bourk*, 6:9-11 (“The electric signal from the microphone 18 is amplified by mic amp 60 and provided to noise cancellation signal processing unit 61.”); 2:65-3:3 (“The above described problems are overcome by a headset for acoustic production of an electronic signal which electronic signal is representative of either an anti-noise signal or the summation of a desired audio signal and an anti-noise signal.”) And like the ’493 patent, *Bourk*’s device is designed to provide audio to a user. See, e.g., ’493 patent, 13:58-62; *Bourk*, Abstract, 2:65-3:3.

250. In my opinion, in view of *Bourk*, a person having ordinary skill in the art would have found applying a low pass filter to *Epley*’s signal from microphone 76 with an upper cutoff frequency less than or equal to 200 Hz amounts to a simple substitution of known elements according to known techniques to yield predictable results (that is, applying a low pass filter to the microphone signal—as already suggested in *Padhi*—with an upper cutoff frequency less than or equal to 200 Hz—as suggested in *Bourk*).

2. **Claim 9: “The audio device according to claim 8, wherein the modified first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200 Hz.”**

251. In my opinion, *Epley* in view of *Padhi*, *Voix*, and *Bourk* discloses or suggests “[t]he audio device according to claim 8, wherein the modified first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200 Hz.”

252. In my opinion, *Bourk* discloses or suggests applying a low pass filter to a modified signal that includes a microphone signal, wherein the upper cutoff frequency is less than or equal to 200 Hz.

253. Specifically, *Bourk* discloses generating a modified signal that is provided to driver 16 (i.e., a loudspeaker) for acoustical reproduction by a headset. *Bourk* generates the modified signal by summing an audio signal (if any) and an anti-noise signal generated based on a signal from **microphone 18**. *Id.*, 4:7-16. *Bourk* discloses that “noise cancellation operates whether or not any such audio signal is provided,” such that the modified signal provided to driver 16 may consist solely of the anti-noise signal generated from **microphone 18**. *Id.*, 5:45-49; *id.*, 6:12-17.

As indicated above, the signal generated by **microphone 18** is provided to electronic circuit 24, at input 26. Electronic circuit 24 processes the desired audio signal provided at input 28, and **generates a modified signal** for acoustical reproduction by the headset. The **modified signal is generated by summing the original audio signal (if any) and an anti-noise signal that is generated based on the signal obtained from microphone 18**. The modified signal is thereafter provided to driver 16.

*Bourk, 4:7-16.*

As indicated above, the electronic signal provided to driver 16 is **generated by summing the anti-noise signal (derived from the microphone signal) with the desired audio signal (if any—the noise cancellation operates whether or not any such audio signal is provided)**.

*Bourk, 5:45-49.*

The **resulting noise signal is then inverted to provide an anti-noise signal that may be amplified and provided to the driver** to cancel the detected noise in the headset. This cancellation is effective whether or not an additional radio communication signal (or similar signal) is also provided to the driver.

*Bourk, 6:12-17.*

254. The modified signal may be filtered before it is provided to driver 16.

*Id.*, 4:23-27; *id.*, 6:25-35. The filters may include, e.g., a high pass filter 62 for

filtering components below a first frequency and a **low pass filter 64 for filtering components above a second frequency**. *Id.*, 6:37-42. *Bourk* discloses that in the preferred embodiment, the first frequency is in the 17 to 70 Hz range and the **second frequency is a range from 150 Hz to 8 kHz**. *Id.*, 6:42-59.

As will be more particularly described in relation to FIG. 4, the ***modified signal may be passed through a plurality of cascaded filters*** for accentuating a desired frequency range prior to being provided to driver 16.

*Bourk*, 4:23-27.

To this end the ***modified audio signal, generated by the summation with the anti-noise signal, may be further processed*** before it is amplified and presented to driver 16. The ***modified audio signal may be processed by its passage through a plurality of cascaded filters*** designed to accentuate a desired frequency range while minimizing undesirable phase characteristics. The exact parameter values of each filter are dependent upon the particular driver, microphone and other acoustic elements chosen for a given headset system.

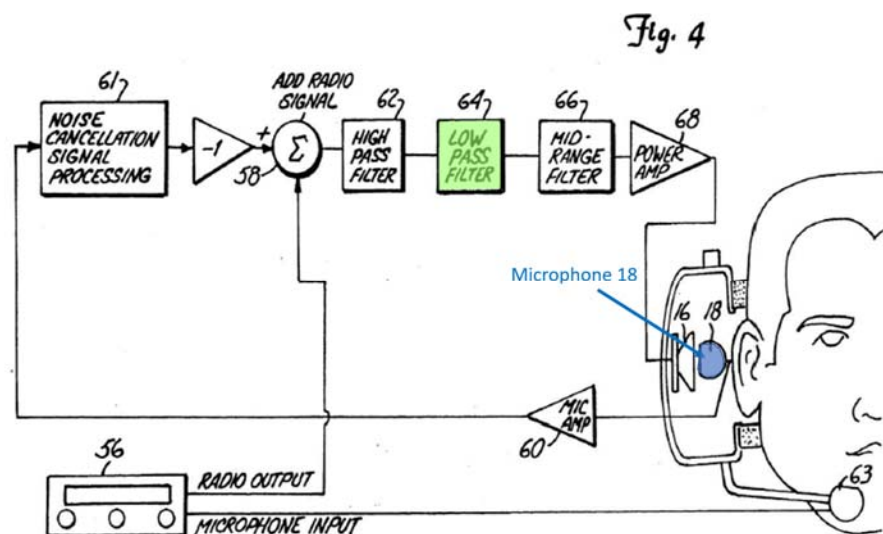
*Bourk*, 6:25-35.

As shown in FIG. 4, the ***series of cascaded filters may include*** a high pass filter 62 for filtering out those components of the modified audio signal having frequencies below a first frequency, and ***a low pass filter 64 for filtering out those components of the modified audio signal having frequencies above a second frequency***.

*Bourk*, 6:37-42.

In the preferred embodiment the first frequency is in a range extending from 17 to 70 Hz and the ***second frequency is also in a range extending from 150 Hz to 8 kHz***. It should be understood that in filter 62 frequency components of the modified audio signal falling below 17 Hz will be attenuated by a fixed amount and that attenuation of the modified audio signal will decrease linearly for frequency components falling between 17 Hz to 70 Hz at which no attenuation takes place. Similarly, in filter 64 frequency components of the modified audio signal falling above 8 kHz will be attenuated by a fixed amount and that attenuation of the modified audio signal increases linearly for frequency components falling between 150 Hz and 8 kHz at which maximum attenuation takes place. The 8 kHz value, a so-called pole, was chosen for the preferred embodiment to minimize phase lag around the frequency of 2.5 kHz.

Bourk, 6:42-59.



Bourk, Fig. 4.

255. In my opinion, a person having ordinary skill in the art would have understood *Bourk* to disclose or suggest choosing an upper cutoff frequency anywhere in the range of 150 Hz to 8 kHz depending on design needs. Further, in my opinion, a person having ordinary skill in the art would have understood *Bourk* to suggest choosing an upper cutoff frequency of 200 Hz because *Bourk* discloses that is the upper end of the range where “maximum noise reduction” will occur. *Bourk*, 7:4-17.

As a result of the above described invention, back ground or ambient noise reduction within the earcup as a function of frequency is predictable from the open loop frequency response of the headset system. ***Maximum noise reduction will occur at frequencies from about 30 Hz to about 200 Hz at values of approximately 15 to 28 dB.*** Above and below the 30 to 200 Hz frequency range, noise reduction falls off according to characteristics determined by the selection of filter parameters. Conventional passive noise reduction built into the headset, however, provides fairly effective noise reduction at higher frequencies, making the head set quite effective in achieving the desired overall reduction of noise, particularly for aircraft applications.

*Bourk*, 7:4-17.

256. As discussed in §IX.B.1, a person having ordinary skill in the art would have been motivated to apply *Bourk*’s disclosure of a “low pass filter” to the

microphone signal wherein the upper cutoff is “200 Hz” in implementing *Epley*’s seal assessment (as modified by *Padhi*’s signal comparison).

257. Thus, in the *Epley* in view of *Padhi*, *Voix*, and *Bourk*, *Epley*’s signal from microphone 18 would first be filtered with a low pass filter with an upper cutoff frequency of, for example, 200 Hz, before it is supplied to *Padhi*’s **cross correlation function 134**.

258. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Epley* in view of *Padhi*, *Voix* and *Bourk* to disclose or suggest that microcontroller 54 is configured to execute instructions stored in ROM to calculate a value that is the result of a non-phase cross correlation calculation ( $\alpha$ ) using the known sound and “a modified” signal from microphone 18 (“wherein the first signal is the test signal and the second signal is a modified first microphone signal”) as required by claim 8. Further, in my opinion, a person having ordinary skill in the art would have understood *Epley* in view of *Padhi*, *Voix* and *Bourk* to disclose or suggest that the “modified” signal from microphone 18 is generated by applying a low pass filter with an upper cutoff frequency that is less than or equal to 200 Hz and may be as low as 150 Hz (“wherein the modified first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200 Hz”).

**3. Claim 12: “The audio device according to claim 11, wherein the first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200 Hz.”**

259. In my opinion, *Epley* in view of *Padhi*, *Voix*, and *Bourk* discloses or suggests “[t]he audio device according to claim 11, wherein the first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200 Hz.”

260. As I described with respect to claim 9, *Bourk* discloses or suggests applying a low pass filter to a modified signal generated based on a microphone signal, wherein the upper cutoff frequency is less than or equal to 200 Hz. See §IX.B.2. Further, as discussed in §IX.B.1, a person having ordinary skill in the art would have been motivated to apply *Bourk*’s disclosure of a “low pass filter” to the microphone signal wherein the upper cutoff is “150 Hz” in implementing *Epley*’s seal assessment (as modified by *Padhi*’s signal comparison).

261. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Epley* in view of *Padhi*, *Voix* and *Bourk* to disclose or suggest that microcontroller 54 is configured to execute instructions stored in ROM to calculate a value that is the result of a non-phase cross correlation calculation ( $\alpha$ ) using the known sound and “a modified” signal from microphone 18 (“wherein the first signal is the test signal and the second signal is a modified

first microphone signal”) as required by claim 8. Further, in my opinion, a person having ordinary skill in the art would have understood *Epley* in view of *Padhi*, *Voix* and *Bourk* to disclose or suggest that the “modified” signal from microphone 18 is generated by applying a low pass filter with an upper cutoff frequency that is less than or equal to 200 Hz and may be as low as 150 Hz (“wherein the modified first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200 Hz”).

**C. Ground 3: *Epley*, *Padhi*, *Voix*, and *Hayashi* Render Obvious Claim 2**

**1. Claim 2: “The audio device according to claim 1, wherein the message is an acoustic message.”**

262. In my opinion, *Epley* in view of *Padhi*, *Voix*, and *Hayashi* discloses or suggests “[t]he audio device according to claim 1, wherein the message is an acoustic message.”

263. In my opinion, to the extent the Board finds that seal quality may be a binary result, *Epley*’s warning is a message conveying seal quality, as I explain in connection with elements [1.2.7] and [1.2.8].

264. Moreover, *Hayashi* discloses an acoustic message. *Hayashi* is titled “Leak tester of ear-mounted external sound processing device.” *Hayashi*, Abstract; [0004]. *Hayashi*’s device functions by “measuring and displaying a leakage

amount of an external sound.” *Hayashi*, [0020]. *Hayashi* discloses “a method may be used to notify the user of the leakage amount of the external sound by **a sound**, vibration, or the like.” *Hayashi*, [0015].

265. A person of ordinary skill in the art would have been motivated to apply *Hayashi*’s teachings of notifying the user of seal quality using sound in implementing *Epley*’s warning to the operator in order to better ensure the user notices the warning. Similar to *Epley*, *Padhi*, and *Voix*, *Hayashi* is analogous art, in the same field as the ’493 patent, and reasonably pertinent to the alleged problems addressed by the ’493 patent of comparing signals for audio applications. *E.g.*, ’493 patent, Abstract (“audio processing circuitry”), 5:17-22; 6:43-45, 10:26-29; *Hayashi*, Abstract, [0001], [0002], [0004], [0013], [0020]. For example, *Hayashi* discloses its seal test involves subtracting an internal microphone signal from an external microphone signal, and comparing the result to a threshold. *Hayashi*, [0013]. And like the ’493 patent, *Hayashi* is directed to a device designed to assess the seal of an audio device. For example, *Hayashi* discloses that “[t]he present invention relates to a device that displays a leakage amount of sound of a hearing aid-type external sound processing device.” *Hayashi*, [0001].

266. *Hayashi* teaches that many different mechanisms can be used to notify a user of seal quality, including using an LED display or “a sound.” *Hayashi*, [0014] (“The display part 20 is composed of the LED 21, and the LED 21 is turned

on based on the input signal from the signal comparison part 10. Note that, in the present embodiment, display is performed using the LED 21, but a method may be used to ***notify the user of the leakage amount of the external sound by a sound, vibration, or the like.***”). Moreover, **Epley** discloses that its “display [that] produces a warning to the operator” is part “of a desktop or portable laptop computer 40.” *Epley*, [0029]. A person of ordinary skill in the art would have understood that such a computer would be capable of providing audio warnings, in addition to or alternatively to the displayed warning. For example, *Epley*’s teaches its “warning” would be displayed on a screen, but an operator of the device may not notice the “warning” immediately. Including an audible sound to indicate seal quality would have attracted the operator’s attention, even if the operator were not viewing the screen at that moment. Further, to the extent it is determined the seal quality is non-binary, a person of ordinary skill in the art would have been motivated to use a non-binary audio warning, such as a sound with multiple tones. See Ground 1 [1.7].

267. A person of ordinary skill in the art would have had a reasonable expectation of success using *Hayashi*’s sound warning in implementing *Epley*’s device. *Epley* device warns its user of seal quality, and using an acoustic sound as part of the warning would have been straightforward and would have required minimal disruption.

268. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Epley* in view of *Padhi*, *Voix*, and *Hayashi* to disclose or suggest that microcontroller 54 is configured to execute instructions stored in ROM to generate and send an audible message to the user that conveys seal quality (“acoustic message”).

**D. Ground 4: *Beard* in view of *Epley*, *Padhi*, and *Voix* (Claims 1, 4, 6, 13-14)**

269. In my opinion, *Beard* in view of *Epley*, *Padhi*, and *Voix* discloses or suggests claims 1, 4, 6, and 13-14.

270. *Beard* describes a personal hearing device designed to provide audio to a user. *Beard*, Abstract.

A ***personal hearing control system*** includes a microphone located within a user's sealed ear canal, a speaker having a diaphragm which directs sound into the ear canal, and a controller which drives the speaker such that the system emulates the acoustics of the user's open ear canal. Also provided are a microphone located on the outer ear side of the sealed ear canal, which is coupled to a handheld interface unit having multiple inputs and operating modes. A user-selected input is processed by the interface unit, and the processed signal is coupled to the speaker. In one operating mode, the output of the outer ear mic is processed so as to cancel sound that leaks from the outer ear to the inner ear side of the seal.

*Beard*, Abstract. *Beard* offers “hearing aids” as an example personal hearing device. *Beard*, [0009].

Various methods have been tried to ameliorate the undesirable distortion caused by the occlusion effect. For example, some ear-sealing **hearing aids** provide small vents between the inside and outside of the ear canal. These vents help, but do not eliminate the effect. Deeply fitted hearing aids exhibit less of the effect, but are uncomfortable and difficult to insert.

*Beard*, [0009]. *Beard* discloses that its device can serve to “correct for hearing deficits in the manner of a hearing aid.” *Beard*, [0047].

The present system provides the user with complete control of his audio input, and not only does not interfere with normal hearing, but can actually enhance it. Furthermore, it can provide hearing protection by limiting the level of sound delivered: for example, interface unit 50 can include a multi-band limiter compressor which can be set to protect the user's hearing and even **correct for hearing deficits in the manner of a hearing aid**, if desired. The system also affords a user privacy, in that only the user can hear what he is listening to.

271. *Beard*'s hearing aid includes “electroacoustic unit containing **speaker 14**, **ear canal microphone 10** and **outer ear microphone 62**.” *Id.*, [0052].

The present personal hearing control system is preferably fitted and calibrated for an individual user. In the preferred embodiment of the earphone element of the system, molds of the user's ears are made

using the same technique used for fitting hearing aids. A soft silicon rubber casting of the user's ear canal, including the entire concha of the external pinna of the ear is made. This casting is then used to make a negative mold which is an accurate replica of the user's ear and ear canal. A soft molding compound such as Locktite Resonaid type 3596 is used to make a soft, snug-fitting ear mold with a cavity into which an **electroacoustic unit containing speaker 14, ear canal microphone 10** and **outer ear microphone 62**, is mounted. The system components which directly interface with the components in the electroacoustic unit are preferably contained in a separate electronics unit (element 140 in FIG. 5, discussed below) located in a headband behind the head that also provides light medial pressure to the ear molds holding them in place. Due to their size and power requirements, the system's digital signal processing electronics are preferably housed within interface unit 50, rather than in the headband portion of the system.

*Beard*, [0052].

272. *Beard* describes an “**ear mold**” that “effectively seal[s] the ear canal.”

*Beard*, [0053].

It is desirable that the **ear mold element of the earphone effectively seal the ear canal**, which serves to prevent higher frequency sounds from reaching the inner ear. However, a very small pressure equalization vent should be provided to equalize the pressure between the sealed ear canal and the outside world, while substantially

attenuating audio frequencies above  $F_{\min}$ . The back volume of the earphone unit which contains the microphones and speaker and acts as the back spring on the speaker diaphragm must also be vented to the outside world by a similar very small vent to provide pressure equalization.

*Beard*, [0053].

273. *Beard* describes that the “**custom ear mold 102**...fits snugly in the concha of the ear...and seals at 107.” *Beard*, [0054].

FIG. 4 is a cut-away drawing showing a preferred configuration for the ear-mounted elements of a system as described herein. Ear canal mic 10, outer ear mic 62 and speaker 14 of the earphone element are contained in a shell 100. The shell is mounted inside a **custom ear mold 102 which fits snugly in the concha of the ear**. Ear mold 102 extends into ear canal 106 and seals at 107. This seal blocks leakage of outside sound and provides for good low frequency response of the system. Seal 107 is preferably designed to leave an area 108 open to the atmosphere, to provide a relief path for pressure that might otherwise build due to mandible vibrations. The back seal volume 110 is preferably standardized, and is the determining contributor to the spring constant experienced by the diaphragm 16 of speaker 14. In practice, this volume should be less than 2 cubic centimeters. The molded earpiece can be designed to be held in place by the external ear itself, or can be held in place by a slight inward pressure by the headband or a spectacle configuration which combines the hearing devices with glasses.

Beard, [0054]. Figure 4 illustrates the **ear mold 102** snugly in the ear.

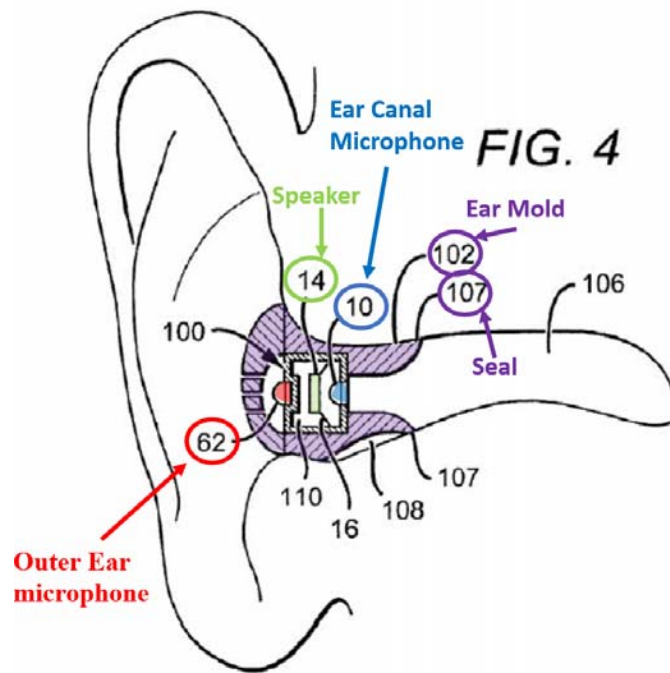


Fig. 4

274. *Beard* describes the benefit of “greater control over what the user hears” when the ear canals are sealed. *Id.*, [0025].

Also note that the present system could be useful even if arranged such that the ear canals are not sealed when used. However, ***greater control over what a user hears is obtained when the system is arranged to seal the ear canals. For this reason, a system which seals both ear canals and controls the sound delivered to both ears is preferred, and is assumed throughout the discussion below.***

*Beard*, [0025].

275. *Beard* discloses “chirps, pseudorandom noise or swept sound signals can be used to determine the impulse transfer functions of the system,” implemented using DSPs. *Beard*, [0044].

In practice, ***chirps, pseudorandom noise or swept sound signals can be used to determine the impulse transfer functions of the system.*** A feedforward noise cancellation scheme of this sort works effectively at lower frequencies, and since it is a feed forward system, there are no stability issues. ***Schemes of this type are typically implemented using one or more digital signal processors (DSPs).*** Methods for implementing filters based on system transfer functions as described herein are well known in the art.

*Beard*, [0044].

276. *Beard* explains “the digital signal processing electronics are preferably housed within **interface unit 50.**” *Beard*, [0052]

The present personal hearing control system is preferably fitted and calibrated for an individual user. In the preferred embodiment of the earphone element of the system, molds of the user's ears are made using the same technique used for fitting hearing aids. A soft silicon rubber casting of the user's ear canal, including the entire concha of the external pinna of the ear is made. This casting is then used to make a negative mold which is an accurate replica of the user's ear and ear canal. A soft molding compound such as Locktite Resonaid type 3596 is used to make a soft, snug-fitting ear mold with a cavity into which an electroacoustic unit containing speaker 14, ear canal

microphone 10, and outer ear microphone 62, is mounted. The system components which directly interface with the components in the electroacoustic unit are preferably contained in a separate electronics unit (element 140 in FIG. 5, discussed below) located in a headband behind the head that also provides light medial pressure to the ear molds holding them in place. Due to their size and power requirements, the system's **digital signal processing electronics are preferably housed within interface unit 50**, rather than in the headband portion of the system.

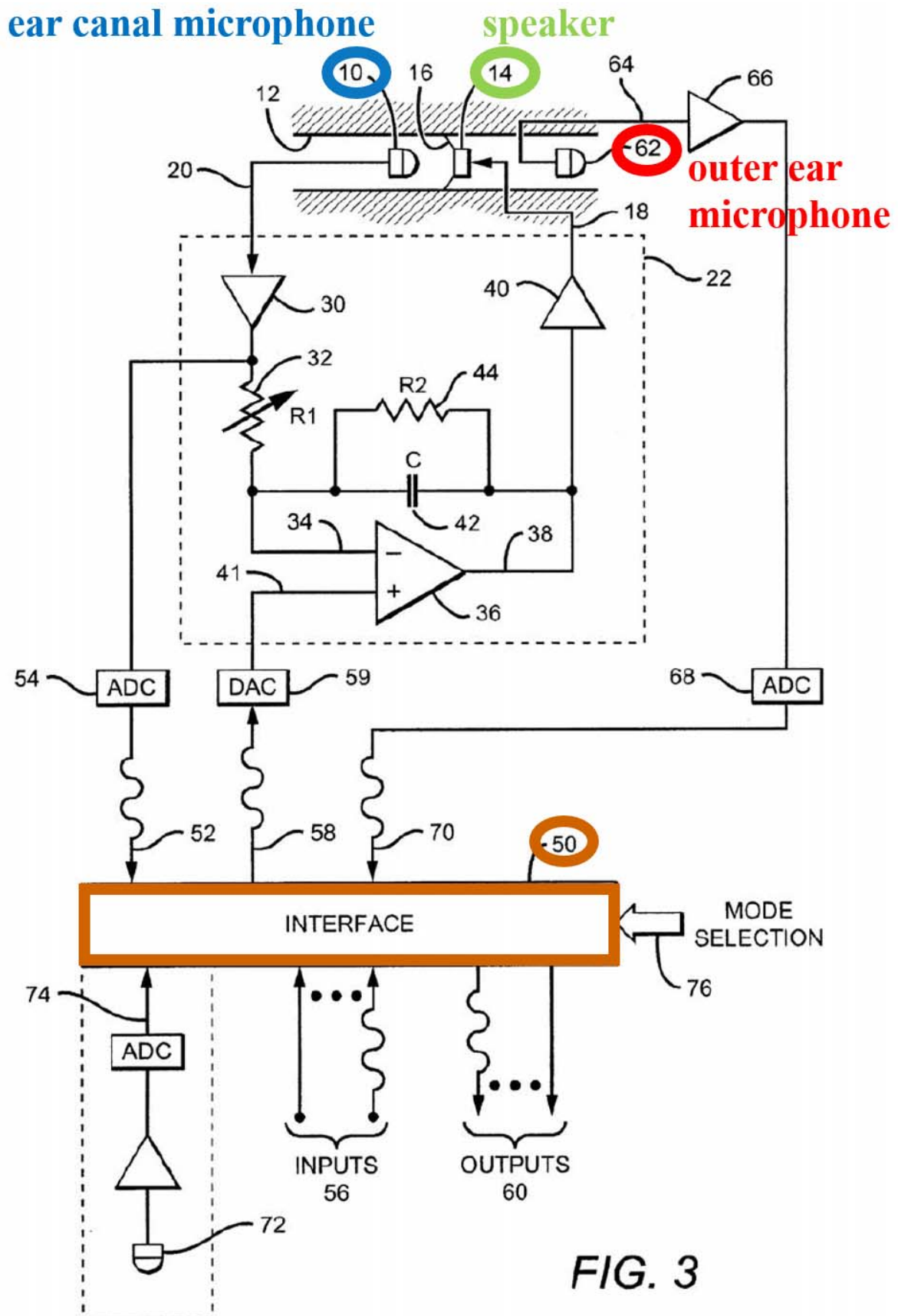
*Beard*, [0052]

277. *Beard* discloses that “a behind-the-head electronics module 140...preferably contains the electronics of the earphone unit,” and is coupled to “electroacoustic modules 122 and 124.” *Beard*, [0059].

FIG. 5 depicts one possible embodiment of a headband 120, which includes the system's earphone components and electronics which are not contained within interface unit 50. An electroacoustic module 122 holds outer ear microphone 62 and ear canal microphone 10 (not shown), and speaker 12 (not shown) for the left ear. A similar module 124 is shown for the right ear. Electroacoustic module 122 fits into a mating cavity 126 in the left ear custom-cast ear mold 128. Module 124 fits into a mating cavity in the right ear custom-cast ear mold 130. Decorative baffles 132 and 134 are mounted on the outside of electroacoustic modules 122 and 124; these baffles refine the acoustic character of the modules. The

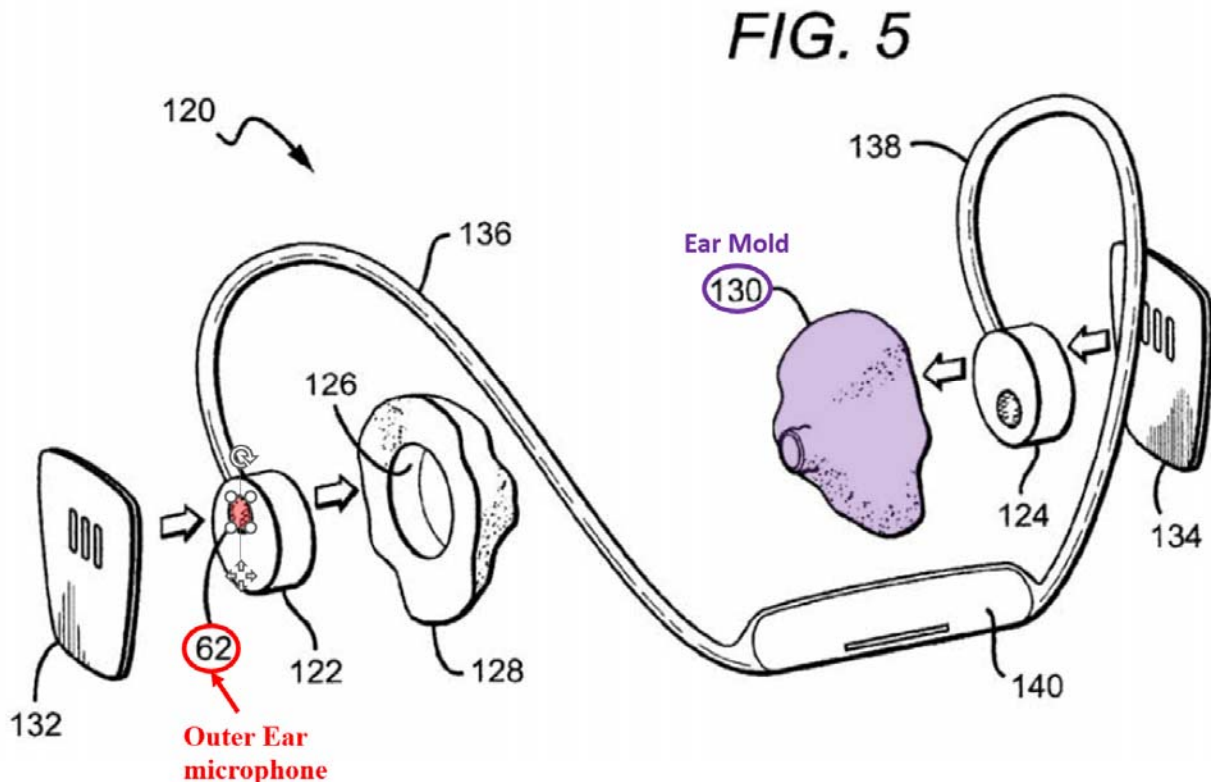
electroacoustic modules are typically cylinders 10-15 millimeters in diameter and 8-10 millimeters high. Headband ear shafts 136 and 138 connect *electroacoustic modules 122 and 124* to a *behind-the-head electronics module 140, which preferably contains the electronics of the earphone unit*—such as op amp 36, buffer amps 30 and 66, driving amp 40, resistors 32 and 44, capacitor 42, ADCs 54 and 68 and DAC 59, as well as circuitry required to implement wireless communication with interface unit 50 as needed—and a rechargeable battery power supply. Ear shafts 136 and 138 are flexible, such that they can be flexed to allow putting the earphones on the user, and flex back to provide inward pressure on the electroacoustic modules helping to hold them and the ear molds in place in the user's ears. Ear molds 128 and 130 may be first inserted into the user's ears, and the headband unit then fitted over the ears and the electroacoustic modules pressed into their mating cavities 126 in the ear molds. The headband units can be manufactured in a range of sizes to comfortably fit users with different size heads and ear placements just as shoes are manufactured in a range of sizes; these headband units would then fit into custom-made ear molds. Electroacoustic modules 122 and 124 and electronics module 140 can be considered standard units amenable to mass production.

*Beard*, [0059]. Figure 3 illustrates **interface unit 50**, as coupled to the components of the personal hearing control system, including the **speaker 14**, **ear canal microphone 10** and **outer ear microphone 62**.



*Beard*, Fig. 3.

278. Figure 5 illustrates the **ear mold** and **outer ear microphone 62**.



### 1. Motivation to Apply *Epley*, *Padhi*, and *Voix* to *Beard*

279. In my opinion, a person of ordinary skill in the art would have been motivated and found it obvious and straightforward to apply *Epley*'s seal test, using *Padhi*'s and *Voix*'s teachings, as explained above in Ground 1.

280. Like *Epley*, *Padhi*, and *Voix*, *Beard* is also analogous art, in the same field as the '493 patent, and reasonably pertinent to the alleged problems addressed

by the '493 patent of comparing signals for audio applications and/or audio sealing. For example, the '493 patent describes an “audio processing circuitry:”

A device or a method using the device includes a balloon configured to seal a user's orifice, where the balloon is configured to produce an acoustic seal between a first side and a second side of the balloon in an ear canal. At least a second side of the balloon is fitted into the ear canal. **Audio processing circuitry** produces an audio signal for driving a speaker in the device and to measure sound level using output from the microphone in the device while the speaker is being driven by the audio signal. **The device or method further includes control circuitry to evaluate a seal quality of the device.** Other embodiments are disclosed.

'493 patent, Abstract.

All transducers shown can receive or transmit audio signals to a processor 116 that undertakes **audio signal processing** and provides a transceiver for audio via the wired (wire 118) or a wireless communication path.

'493 patent, 5:17-22.

The RAM 202 stores program instructions for execution on the processor 206 as well as **captured audio processing data**.

'493 patent, 6:43-45.

The cross-correlation estimate using the above recursion is modified for block-wise **processing** rather than the sample-by-sample basis.

'493 patent, 10:26-29.

281. Likewise, Beard discloses processing audio:

A personal hearing control system includes a microphone located within a user's sealed ear canal, a speaker having a diaphragm which directs sound into the ear canal, and a controller which drives the speaker such that the system emulates the acoustics of the user's open ear canal. Also provided are a microphone located on the outer ear side of the sealed ear canal, which is coupled to a handheld interface unit having multiple inputs and operating modes. ***A user-selected input is processed by the interface unit,*** and the processed signal is coupled to the speaker. In one operating mode, the output of the outer ear mic is ***processed*** so as to cancel sound that leaks from the outer ear to the inner ear side of the seal.

*Beard*, Abstract.

The interface unit preferably has multiple selectable operating modes. For example, the interface could be arranged such that the output of the ***outer ear microphone is processed*** such that the signal coupled to the speaker cancels sound that leaks from the outer ear to the inner ear side of the seal. In another operating mode, the system provides accurate high fidelity reproduction by the speaker of sound received by the outer ear microphone.%

*Beard*, [0013].

The preferred embodiment of the present personal hearing control system includes an interface unit having one or more inputs suitable

for connection to respective sources of audio, which is arranged to ***process the audio*** received at a selected input and ***to couple the processed output*** to the speaker. ***The interface unit preferably has selectable operating modes which establish the characteristics of the processed output.*** An exemplary embodiment of a personal hearing control system which includes such an interface unit 50 is shown in FIG. 3. ***As noted above, the interface is arranged to receive and process one or more audio inputs,*** preferably in the digital domain, and provide an output which is coupled to speaker 14. For example, the output of ear canal microphone buffer amplifier 30 could be provided to an interface input 52, preferably after being converted to a digital signal using an analog-to-digital converter (ADC) 54. Inputs 56 might also be provided, for connection to audio sources such as a cell phone, telephone, MP3 player, radio, TV, computer, video game system, closed-circuit theater sound system, etc. An output 58 is preferably coupled to speaker 14 via a digital-to-analog converter (DAC) 59, the analog output of which is fed to the non-inverting input of op amp 36; additional outputs 60 are provided as needed. Some of the interface unit's inputs and outputs are preferably wireless connections, which are depicted as wavy lines in FIG. 3.

*Beard*, [0039].

282. A person of ordinary skill in the art would have been motivated to apply *Epley's* seal assessment teachings, using *Voix's* threshold and frequency range teachings and *Padhi's* cross-correlation teachings (as explained in Ground

1), to *Beard*'s teachings of an interface unit connected to earphones with a "sealed ear canal" because *Beard* leaves it to a person of ordinary skill in the art to decide how to determine whether a sufficient seal exists. While *Beard* discloses "a system which seals both ear canals...is assumed throughout the discussion," *Beard* does not disclose how to determine whether a sufficient seal exists. *Beard*, [0025].

For example;

Also note that the present system could be useful even if arranged such that the ear canals are not sealed when used. However, greater control over what a user hears is obtained when the system is arranged to seal the ear canals. For this reason, ***a system which seals both ear canals and controls the sound delivered to both ears is preferred, and is assumed throughout the discussion below.***

*Beard*, [0025].

283. *Beard* further discloses that it supports a range of modes of operation selectable by the user. *Beard*, [0042].

Interface unit 50 can also include a digital audio memory for storing on board audio information like music. The interface can include digital processing power that allows actively modifying, storing, and sending digital audio information in various forms. ***With these features, the present system can operate in a wide range of modes selectable by the user.*** For instance, assuming the ear canal is substantially sealed by the system, if interface unit 50 sends no data via output 58 to DAC 59, the user will experience silence. That is, the

components mounted in the ear will effectively block outside sound, controller 22 will emulate the acoustics of the user's open ear canal, and the zero input applied at non-inverting input 41 will further attenuate any outside sound leaking into the ear below the critical frequency.¶

*Beard*, [0042].

284. A person of ordinary skill in the art would have understood that a seal test is “applicable to any in-ear devices,” such as hearing aids. *Voix*, [0005].

All the above specifically refer to earplugs but it is to be understood that it is similarly ***applicable to any in-ear device***, the latter referring to either earplug device (or hearing protection device (HPD)) or ***hearing aid device*** (HAD) for which an attenuation level or an amplification performance level is seek respectively.

*Voix*, [0005]. For example, *Svean*, discloses that “[t]he result of leakage is reduced damping of potentially harmful noise. Ideally, the leakage should be detected and remedied prior to noise exposure.” *Svean*, 3:60-64. *Svean* explains this in the context of identifying a number of reasons why a product that is designed to provide a seal may not actually provide a seal when inserted in a user’s ear.

Experience shows that several factors may compromise the sealing of a hearing protector and thereby increase the risk of hearing damage. These factors include

Irregular surfaces which the sealing material is not able to follow properly. Examples are spectacles used with ear cups, and earplugs used by persons with irregularly formed ear canals.

Improper placement of the hearing protector. Experience and patience is required by the user to get a hearing protector mounted correctly. In cases where the user is wearing a helmet or cap, the hearing protector may be accidentally pushed out of position during use.

Ageing of the materials in the sealing may reduce the resilience of the sealing and thereby allow leakage around the sealing.

The result of leakage is reduced damping of potentially harmful noise. Ideally, the leakage should be detected and remedied prior to noise exposure. The leakage may not be clearly audible. Accordingly, noise situations may comprise of intermittent or impulsive components which may damage the hearing almost instantaneously if a hearing protector should be malfunctioning or imperfect without the user's knowledge.

*Svean*, 3:44-66. Thus, even though *Beard* assumes a proper seal is in place, a person of ordinary skill in the art would have understood that a seal test is necessary to ensure that an actual seal exists. *Epley*, as modified by *Padhi* and *Voix*, provides such a seal test.

285. Further, *Beard*'s interface unit 50 is coupled to a "computer," and "additional outputs 60 are provided as needed." *Beard* [0039]. A person of ordinary skill in the art would have understood that, in applying *Epley*'s teachings

of a seal test to *Beard, Epley's* teachings of transmission of a “warning” message to a “display” of a computer would be achieved by sending a warning to *Beard's* computer via one of the “additional outputs 60.”

The preferred embodiment of the present personal hearing control system includes an interface unit having one or more inputs suitable for connection to respective sources of audio, which is arranged to process the audio received at a selected input and to couple the processed output to the speaker. The interface unit preferably has selectable operating modes which establish the characteristics of the processed output. An exemplary embodiment of a personal hearing control system which includes such an **interface unit 50** is shown in FIG. 3. As noted above, the interface is arranged to receive and process one or more audio inputs, preferably in the digital domain, and provide an output which is coupled to speaker 14. For example, the output of ear canal microphone buffer amplifier 30 could be provided to an interface input 52, preferably after being converted to a digital signal using an analog-to-digital converter (ADC) 54. Inputs 56 might also be provided, for connection to audio sources such as a cell phone, telephone, MP3 player, radio, TV, **computer**, video game system, closed-circuit theater sound system, etc. An output 58 is preferably coupled to speaker 14 via a digital-to-analog converter (DAC) 59, the analog output of which is fed to the non-inverting input of op amp 36; additional **outputs 60 are provided as needed**. Some of the interface unit's inputs and outputs are

preferably wireless connections, which are depicted as wavy lines  
in FIG. 3.

*Beard*, [0039].

When ***properly arranged***, the present hearing control system provides the capability of environmental noise control, private communication, and a practical and effective platform technology for recording and playing back 3D audio, with the handheld interface unit controlling the mode of operation for various communication, entertainment and listening applications.

*Beard*, [0014]. Thus, a person of ordinary skill in the art would have been motivated to apply *Epley*'s seal test (including its transmission of a "warning" message to a "display" of a computer, *see* ground 1) to *Beard*'s interface unit 50 coupled to a "computer" to inform the user of "proper[] arrange[ment]." *Beard*, [0011], [0039].

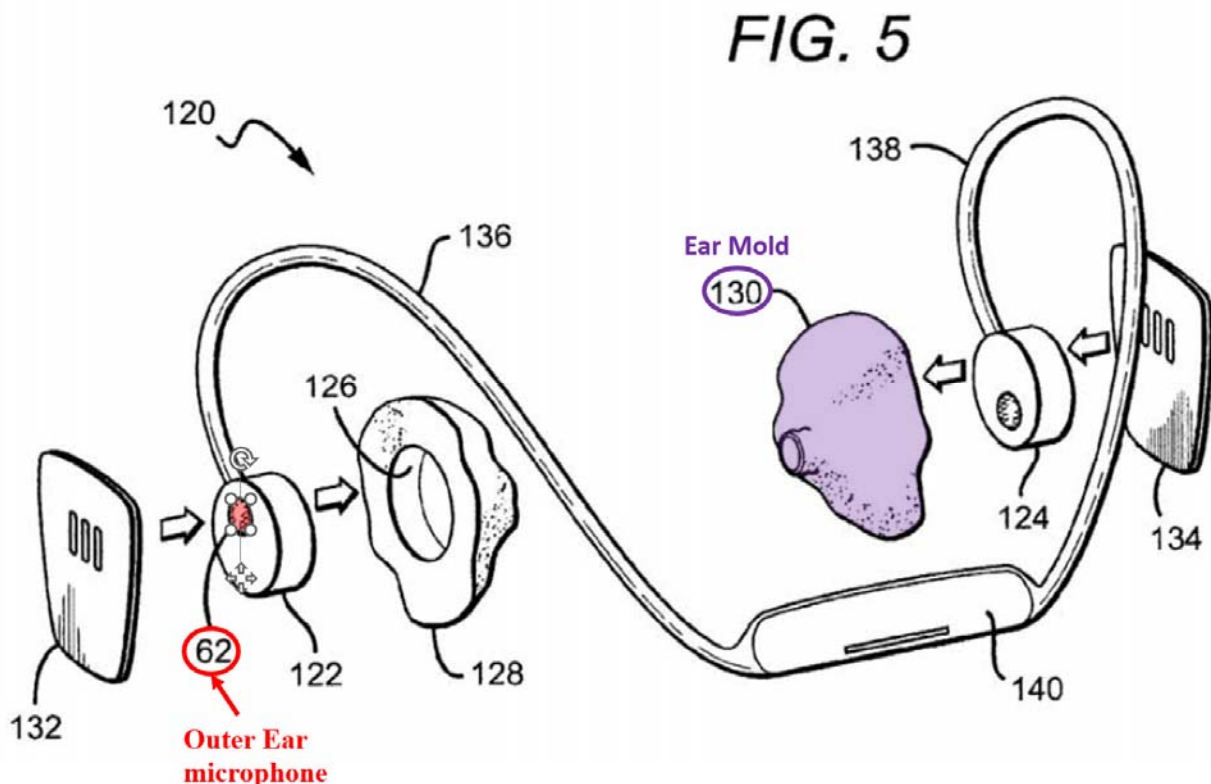
286. It is my opinion that a person of ordinary skill in the art would have had a reasonable expectation of success applying the teachings of *Epley*'s seal assessment, using *Voix*'s threshold and frequency range and *Padhi*'s cross-correlation (*see* ground 1), in implementing *Beard*'s electronics module. *Beard* teaches its "interface unit 50" is connected to the "electroacoustic modules" in a similar fashion to which *Epley*'s "feedback mechanism" is "operatively coupled" to "ear probes". *Beard*, [0059]; *Epley*, [0019]. For example, *Epley* discloses that

this operative coupling is achieved through a “wireless” connection, similar to *Beard*’s wireless connection between “interface unit 50” and the “electroacoustic modules” via “a behind-the-head electronics module 140.” *Beard*, [0059], Fig. 5; *Epley*, [0019].

FIG. 5 depicts one possible embodiment of a headband 120, which includes the system's earphone components and electronics which are not contained within interface unit 50. An electroacoustic module 122 holds outer ear microphone 62 and ear canal microphone 10 (not shown), and speaker 12 (not shown) for the left ear. A similar module 124 is shown for the right ear. Electroacoustic module 122 fits into a mating cavity 126 in the left ear custom-cast ear mold 128. Module 124 fits into a mating cavity in the right ear custom-cast ear mold 130. Decorative baffles 132 and 134 are mounted on the outside of electroacoustic modules 122 and 124; these baffles refine the acoustic character of the modules. The electroacoustic modules are typically cylinders 10-15 millimeters in diameter and 8-10 millimeters high. Headband ear shafts 136 and 138 ***connect electroacoustic modules 122 and 124 to a behind-the-head electronics module 140, which preferably contains the electronics of the earphone unit—such as op amp 36, buffer amps 30 and 66, driving amp 40, resistors 32 and 44, capacitor 42, ADCs 54 and 68 and DAC 59, as well as circuitry required to implement wireless communication with interface unit 50 as needed***—and a rechargeable battery power supply. Ear shafts 136 and 138 are flexible, such that they can be flexed to allow

putting the earphones on the user, and flex back to provide inward pressure on the electroacoustic modules helping to hold them and the ear molds in place in the user's ears. Ear molds 128 and 130 may be first inserted into the user's ears, and the headband unit then fitted over the ears and the electroacoustic modules pressed into their mating cavities 126 in the ear molds. The headband units can be manufactured in a range of sizes to comfortably fit users with different size heads and ear placements just as shoes are manufactured in a range of sizes; these headband units would then fit into custom-made ear molds. Electroacoustic modules 122 and 124 and electronics module 140 can be considered standard units amenable to mass production.

*Beard*, [0059].



*Beard*, Fig. 5.

System 10 further includes an ocular analyzer, e.g. nystagmus analyzer 20, operatively coupled to camera 14, a head-motion analyzer 22 operatively coupled to positional/inertial sensor 16 and a ***sound and pressure generator 24 operatively coupled to one or more and preferably bilateral ear probes 18***. Those of skill in the art will appreciate that, ***within the spirit and scope of the invention, such operative coupling can be physical or wireless***, e.g. telemetric.

Epley, [0019].

287. Moreover, *Beard* teaches the DSP of its “interface 50” assesses the earphone functionality, including the “impulse transfer functions of the system.”

*Beard*, [0039], [0044].

The preferred embodiment of the present personal hearing control system includes an ***interface unit having one or more inputs suitable for connection*** to respective sources of audio, which is arranged to process the audio received at a selected input and to couple the processed output to the speaker. ***The interface unit preferably has selectable operating modes which establish the characteristics of the processed output***. An exemplary embodiment of a personal hearing control system which includes such an interface unit **50** is shown in FIG. 3. As noted above, the interface is arranged to receive and process one or more audio inputs, preferably in the digital domain, and provide an output which is coupled to speaker 14. For example, the output of ear canal microphone buffer amplifier 30 could be

provided to an interface input 52, preferably after being converted to a digital signal using an analog-to-digital converter (ADC) 54. Inputs 56 might also be provided, for connection to audio sources such as a cell phone, telephone, MP3 player, radio, TV, computer, video game system, closed-circuit theater sound system, etc. An output 58 is preferably coupled to speaker 14 via a digital-to-analog converter (DAC) 59, the analog output of which is fed to the non-inverting input of op amp 36; additional outputs 60 are provided as needed. Some of the interface unit's inputs and outputs are preferably wireless connections, which are depicted as wavy lines in FIG. 3.

*Beard*, [0039].

In practice, chirps, pseudorandom noise or swept sound signals can be used to determine the ***impulse transfer functions of the system***. A feedforward noise cancellation scheme of this sort works effectively at lower frequencies, and since it is a feed forward system, there are no stability issues. Schemes of this type are typically implemented using one or more digital signal processors (DSPs). Methods for implementing filters based on system transfer functions as described herein are well known in the art.%

*Beard*, [0044]. Thus, *Beard* recognizes that its device provides functionality beyond a simple hearing device. And like *Epley*, *Beard* teaches (1) a speaker that can produce a known sound inside the ear canal and (2) an ear canal microphone that detects sound inside the ear. *Beard*, [0012], [0026], [0052].

In a preferred embodiment, the personal hearing control system is fitted to seal the user's ear canal, with the ***speaker and ear canal microphone located on the inner ear side of the seal***. The system preferably also includes a microphone located on the outer ear side of the seal, and a handheld interface unit having one or more inputs suitable for connection to respective sources of audio including the outer ear microphone. The interface is arranged to produce an output that varies with the audio received at a selected one of its inputs, and to couple the output—preferably wirelessly—to the speaker.

*Beard*, [0012].

The arrangement of components described above is symbolically illustrated in FIG. 1. ***An 'ear canal microphone' 10 is located within a user's ear canal 12, and a speaker 14 having a diaphragm 16 is arranged to direct sound into the user's ear canal*** in response to a driving signal 18. Ear canal microphone 10 produces an output signal 20 which varies with the instantaneous pressure in the ear canal. A controller 22 receives this signal and produces driving signal 18 so as to control the relationship between the instantaneous pressure in the ear canal and the speaker diaphragm's velocity such that the velocity is proportional to the instantaneous pressure. Thus, when the pressure measured by microphone 10 increases, a force is applied to speaker 14 causing the velocity of diaphragm 16 to increase to the right in FIG. 1. When the relationship between diaphragm velocity and instantaneous air pressure is properly established over the range of sound frequencies that would normally be affected by the

occlusion effect, the system emulates the acoustics of the user's open ear canal, thereby reducing or eliminating the occlusion effect that would otherwise be experienced.

*Beard*, [0026].

The present personal hearing control system is preferably fitted and calibrated for an individual user. In the preferred embodiment of the earphone element of the system, molds of the user's ears are made using the same technique used for fitting hearing aids. A soft silicon rubber casting of the user's ear canal, including the entire concha of the external pinna of the ear is made. This casting is then used to make a negative mold which is an accurate replica of the user's ear and ear canal. A soft molding compound such as Loctite Resonaid type 3596 is used to make ***a soft, snug-fitting ear mold with a cavity into which an electroacoustic unit containing speaker 14, ear canal microphone 10, and outer ear microphone 62, is mounted.*** The system components which directly interface with the components in the electroacoustic unit are preferably contained in a separate electronics unit (element 140 in FIG. 5, discussed below) located in a headband behind the head that also provides light medial pressure to the ear molds holding them in place. Due to their size and power requirements, the system's digital signal processing electronics are preferably housed within interface unit 50, rather than in the headband portion of the system.

*Beard*, [0052].

288. Thus, it would have been straightforward to implement the DSP of *Beard*'s interface 50 by applying the teachings of *Epley*'s sealing assessment software instructions (using the teachings of *Voix*'s threshold and frequency range and *Padhi*'s cross-correlation) to assess the seal quality of the earphones, and a person of ordinary skill in the art would have known that such a combination (yielding the claimed limitations) would predictably work and provide the expected functionality.

## 2. Claim 1

### a. [1.pre]: “An audio device configured to test a seal of an earphone comprising:”

289. I understand that “[a]n audio device configured to test a seal of an earphone comprising” is the preamble to claim 1 of the '493 patent. I have been asked to assume that the preamble is a claim limitation. It is my opinion that *Beard* discloses or suggests this feature.

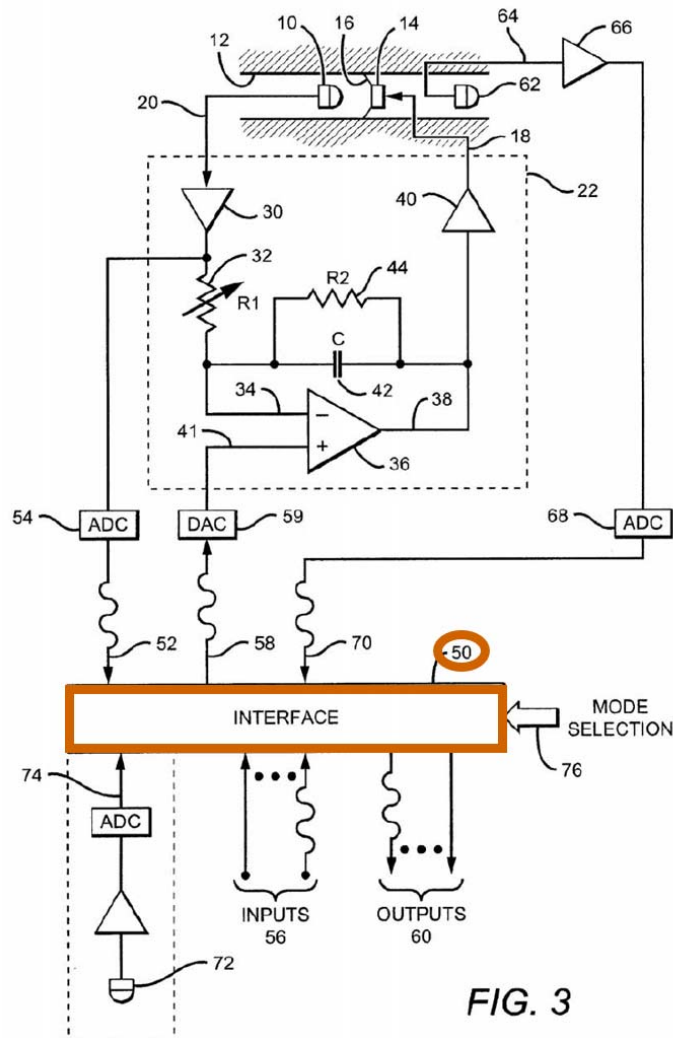
290. *Beard*'s “interface unit 50” corresponds to the claimed “audio device” and the “earphone” has a seal corresponding to “a seal of an earpiece.” *Beard*, [0012], [0054]. *Beard*'s “**interface unit 50**...includes multiple operating modes...[that] dictate[] which input signal is selected and how the selected signal is processed.” *Beard*, [0041]. Input can be “modified in accordance with the selected operating mode.” *Beard*, [0041].

***Interface unit 50 preferably includes multiple operating modes***, with a means 76 provided with which a desired mode can be selected. ***The selected operating mode dictates which input signal is selected and how the selected signal is processed*** before being coupled to speaker 14. ***A selected input can be*** equalized, compressed, limited or otherwise ***modified in accordance with the selected operating mode***.

*Beard*, [0041]. *Beard's interface unit 50* includes memory for “storing on board audio information.” *Beard*, [0042].

***Interface unit 50*** can also include a digital audio memory for ***storing on board audio information*** like music. The interface can include digital processing power that allows actively modifying, storing, and sending digital audio information in various forms. With these features, the present system can operate in a wide range of modes selectable by the user. For instance, assuming the ear canal is substantially sealed by the system, if interface unit 50 sends no data via output 58 to DAC 59, the user will experience silence. That is, the components mounted in the ear will effectively block outside sound, controller 22 will emulate the acoustics of the user's open ear canal, and the zero input applied at non-inverting input 41 will further attenuate any outside sound leaking into the ear below the critical frequency.

*Beard*, [0042]. *Beard* illustrates ***interface unit 50*** in figure 3, below.



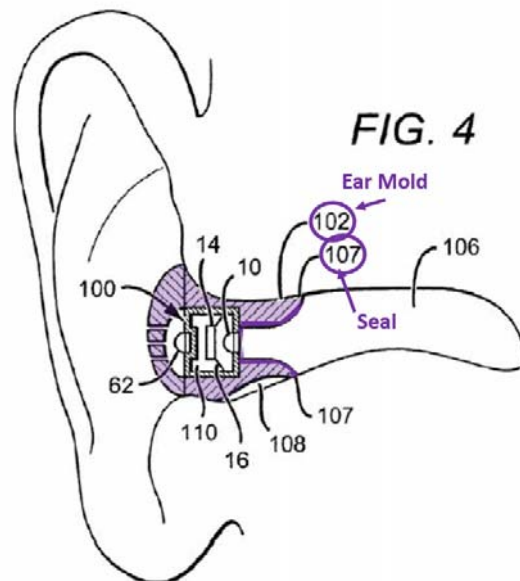
Beard, Fig. 3

291. *Beard* also describes an “earphone” with “[e]ar mold 102 [that] extends into ear canal 106 and seals at 107.” *Beard*, [0054].

FIG. 4 is a cut-away drawing showing a preferred configuration for the ear-mounted elements of a system as described herein. Ear canal mic 10, outer ear mic 62 and speaker 14 of the *earphone* element are contained in a shell 100. The shell is mounted inside a *custom ear mold 102 which fits snugly in the concha of the ear. Ear mold 102*

***extends into ear canal 106 and seals at 107.*** This seal blocks leakage of outside sound and provides for good low frequency response of the system. Seal 107 is preferably designed to leave an area 108 open to the atmosphere, to provide a relief path for pressure that might otherwise build due to mandible vibrations. The back seal volume 110 is preferably standardized, and is the determining contributor to the spring constant experienced by the diaphragm 16 of speaker 14. In practice, this volume should be less than 2 cubic centimeters. The molded earpiece can be designed to be held in place by the external ear itself, or can be held in place by a slight inward pressure by the headband or a spectacle configuration which combines the hearing devices with glasses.

*Beard*, [0054]. *Beard* illustrates **ear mold 102** in figure 4, below.



*Beard*, Fig. 4

292. Thus, a person of ordinary skill in the art would have understood that Beard's "**interface unit 50**" and "earphone" can be configured in an "operating mode" to test the seal of "**ear mold 102**."

293. Moreover, as I discussed above in Ground 1, Epley discloses [1.pre], to the extent that the preamble is limiting. As I discussed in Section IX.D.1, a person of ordinary skill in the art would have been motivated to apply *Epley*'s seal assessment to *Beard*'s personal hearing system with a sealed ear canal.

**b. [1.1]: "a memory that stores instructions; and"**

294. In my opinion, *Beard* discloses or suggests "a memory that stores instructions."

295. *Beard* discloses "**interface unit 50**" includes "digital audio memory" and "digital processing power that allows actively modifying, storing, and sending digital audio information." The use of the digital processing allows the device to "operate in a wide range of modes selectable by the user." *Beard*, [0042].

**Interface unit 50** can also *include a digital audio memory* for storing on board audio information like music. The interface can include *digital processing power that allows actively modifying, storing, and sending digital audio information* in various forms. With these features, the present system can *operate in a wide range of modes selectable by the user*. For instance, assuming the ear canal is

substantially sealed by the system, if interface unit 50 sends no data via output 58 to DAC 59, the user will experience silence. That is, the components mounted in the ear will effectively block outside sound, controller 22 will emulate the acoustics of the user's open ear canal, and the zero input applied at non-inverting input 41 will further attenuate any outside sound leaking into the ear below the critical frequency.

*Beard*, [0042]. *Beard's* “selected operating mode dictates...how the selected signal is processed.” [0041].

Interface unit 50 preferably includes **multiple operating modes**, with a means 76 provided with which a desired mode can be selected. **The selected operating mode dictates which input signal is selected** and how the selected signal is processed before being coupled to speaker 14. A selected input can be equalized, compressed, limited or otherwise modified in accordance with the selected operating mode.

*Beard*, [0041].

296. Moreover, the “DSP” implements “noise cancellation scheme.”

*Beard*, [0044].

In practice, chirps, pseudorandom noise or swept sound signals can be used to determine the impulse transfer functions of the system. A feedforward **noise cancellation scheme** of this sort works effectively at lower frequencies, and since it is a feed forward system, there are

no stability issues. Schemes of this type are typically implemented using one or more **digital signal processors (DSPs)**. Methods for implementing filters based on system transfer functions as described herein are well known in the art.

*Beard*, [0044].

297. Thus, a person of ordinary skill in the art would have understood that, for the processor to allow a user to select modes and manipulate audio information, the memory is associated with instructions. At the very least, a person of ordinary skill in the art would have found it obvious that the memory would include instructions to allow the user to select modes and manipulate audio information.

**c. [1.2]: “a processor, wherein the processor is configured to execute the instructions to perform operations, the operations comprising:”**

298. In my opinion, *Beard* discloses or suggests “a processor, wherein the processor is configured to execute the instructions to perform operations, the operations comprising.”

299. *Beard* discloses that “interface 50” includes “digital processing power” that can manipulate information. *Beard*, [0042].

Interface unit 50 can also include a digital audio memory for storing on board audio information like music. ***The interface can include digital processing power that allows actively modifying, storing, and sending digital audio information in various forms. With these features, the present system can operate in a wide range of modes***

***selectable by the user.*** For instance, assuming the ear canal is substantially sealed by the system, if interface unit 50 sends no data via output 58 to DAC 59, the user will experience silence. That is, the components mounted in the ear will effectively block outside sound, controller 22 will emulate the acoustics of the user's open ear canal, and the zero input applied at non-inverting input 41 will further attenuate any outside sound leaking into the ear below the critical frequency.

*Beard*, [0042].

Due to their size and power requirements, the system's digital signal processing electronics are preferably housed within interface unit 50, rather than in the headband portion of the system.

*Beard*, [0052].

300. *Beard's* processors are implemented as DSPs that implement “schemes” like noise cancellation. *Beard*, [0043], [0044].

Further attenuation of outside sound can be achieved by using the signal from outer ear microphone 62 to produce a signal which cancels the audio leaking into the sealed ear canal. This is accomplished by providing one or more ***filters within interface 50 which process the audio signal produced by outer ear microphone 62 such that a signal is coupled to speaker 14 that cancels sound that leaks from the outer ear to the inner ear side of the seal.*** Such a filter is preferably implemented by first ***determining the system transfer function between the audio signal produced by outer ear***

***microphone 62 and the output of ear canal microphone 10, and the system transfer function between speaker 14 and the output of ear canal microphone 10.*** Once known, one or more filters can be implemented based on the transfer functions, which allow for the accurate cancellation of outside sound leaking into the sealed ear canal, as well as accurate high fidelity reproduction by the speaker of sound received by the outer ear microphone. The filters can be, for example, feedforward finite impulse response (FIR) digital filters, with each filter's coefficients calculated based on the system transfer functions.

In practice, chirps, pseudorandom noise or swept sound signals can be used to determine the impulse transfer functions of the system. ***A feedforward noise cancellation scheme of this sort works effectively at lower frequencies, and since it is a feed forward system, there are no stability issues. Schemes of this type are typically implemented using one or more digital signal processors (DSPs).*** Methods for implementing filters based on system transfer functions as described herein are well known in the art.

***The ability to cancel sound that leaks from the outer ear to the inner ear side of the seal is preferably provided as one of the interface unit's operating modes.*** For example, if a user wishes to make a phone call in a noisy environment, he could elect to activate the noise cancellation functionality described above, and thereby enhance his ability to clearly hear the other party.

*Beard*, [0043]-[0045].

301. Moreover, to the extent that “the processor is configured to execute the instructions to perform operations, the operations comprising” is a means-plus-function limitation, *Beard* discloses a processor, as explained above, and as explained below, the processor performs the algorithm disclosed by the ’493 patent.

302. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Beard* to disclose **interface 50** includes a DSP (“a processor”) that implement schemes (“wherein the processor is configured to execute the instructions to perform operations”).

**d. [1.2.1]: “generating a test signal,”**

303. In my opinion, *Beard* in view of *Epley* discloses or suggests “generating a test signal.”

304. As I explain above in connection with element [1.2], *Beard* discloses “schemes” implemented by a “DSP.”

305. *Beard* discloses generating an audio signal. For example, *Beard* discloses its DSP generates “swept sound signals” for calibrating feed forward cancellation. *Beard*, [0044]

***In practice, chirps, pseudorandom noise or swept sound signals can be used to determine the impulse transfer functions of the system. A feedforward noise cancellation scheme of this sort works effectively at lower frequencies, and since it is a feed forward system, there are***

no stability issues. ***Schemes of this type are typically implemented using one or more digital signal processors (DSPs).*** Methods for implementing filters based on system transfer functions as described herein are well known in the art.

*Beard*, [0044].

306. Moreover, *Epley* discloses “generating a test signal,” as I explain above in Ground 1. § IX.A.6.d. As I explain above in §IX.D.1, a person of ordinary skill in the art would have been motivated to apply *Epley*’s seal assessment teachings (including test signal generation) to *Beard*’s personal hearing system.

307. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Beard* in view of *Epley* to disclose or suggest instructions executed in the processor for “generating a test signal.”

(a) **[1.2.1.1]: “wherein the test signal includes multiple frequency components, wherein at least one component lies between a range of 30 Hz and 200 Hz;”**

308. In my opinion, *Beard* in view of *Epley* and *Voix* discloses or suggests “wherein the test signal includes multiple frequency components, wherein at least one component lies between a range of 30 Hz and 200 Hz.”

309. As I explain in §IX.A.5, a person of ordinary skill in the art would have been motivated to apply *Voix*’s teachings of a test signal range to *Epley*’s seal

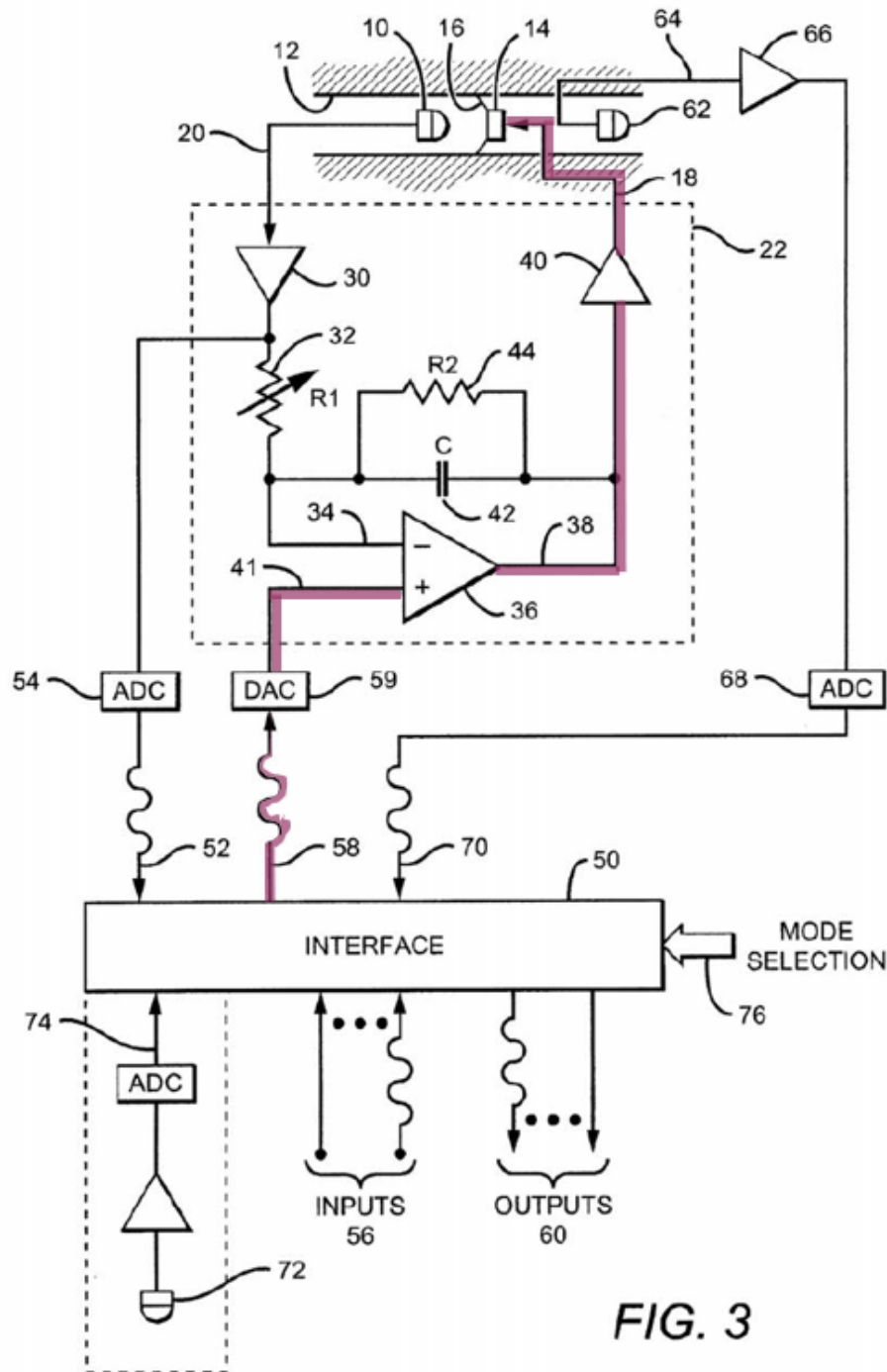
assessment. As discussed in §IX.D.1, a person of ordinary skill in the art would have been motivated to apply *Epley*'s seal assessment teachings (including test signal generation) to *Beard*'s personal hearing system.

- (ii) **[1.2.2]-[1.2.3]: “sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user,”**

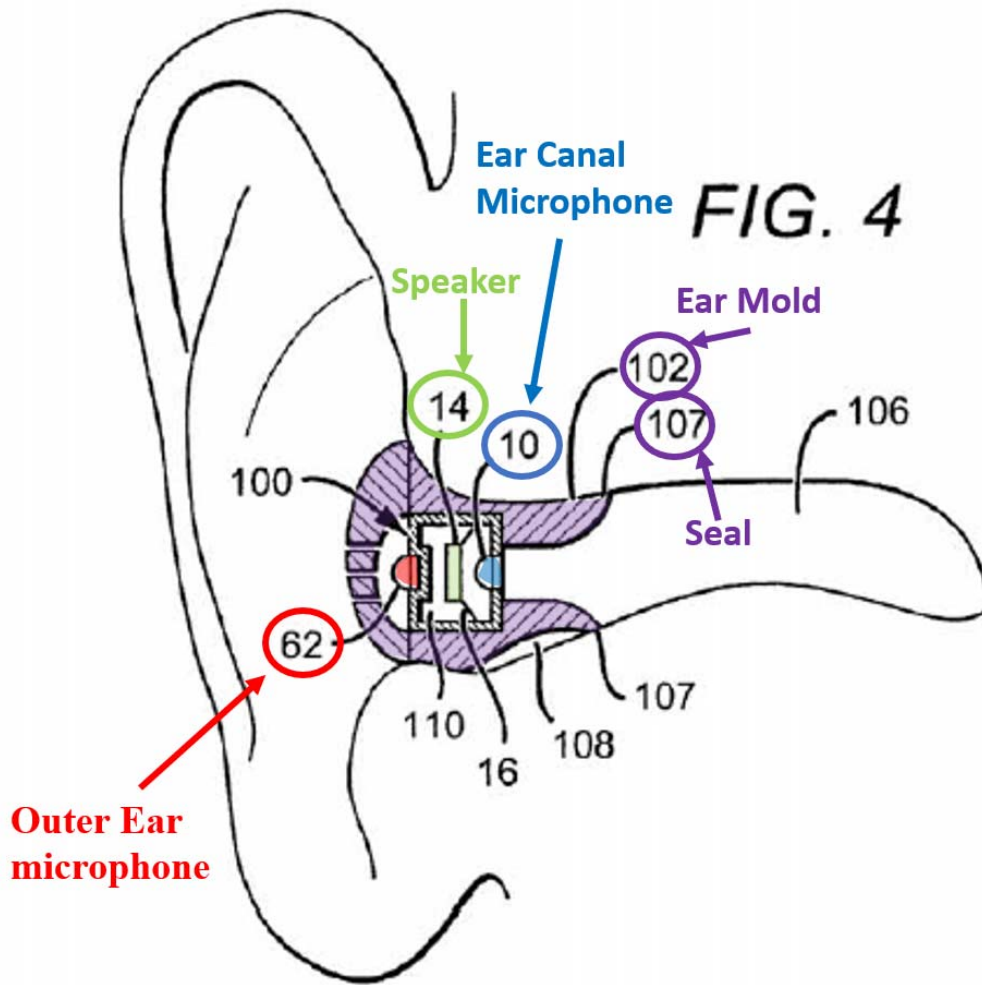
310. In my opinion, *Beard* in view of *Epley* discloses or suggests “sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user.”

311. As I explain above in connection with element [1.2], *Beard* discloses “schemes” implemented by a “DSP.”

312. Moreover, *Beard* discloses sending audio signals from interface 50 to speaker 14 of earphones, as illustrated in Figure 3.



*Beard*, Fig. 3 (annotated). As illustrated in Figure 4, the speaker 14 is part of the earphone.



Beard, Fig. 4 (annotated).

**FIG. 4 is a cut-away drawing showing a preferred configuration for the ear-mounted elements of a system as described herein.** Ear canal mic 10, outer ear mic 62 and **speaker 14 of the earphone element are contained in a shell 100.** The shell is mounted inside a custom ear mold 102 which fits snugly in the concha of the ear. Ear mold 102 extends into ear canal 106 and seals at 107. This seal blocks leakage of outside sound and provides for good low frequency response of the system. Seal 107 is preferably designed to leave an area 108 open to

the atmosphere, to provide a relief path for pressure that might otherwise build due to mandible vibrations.

*Beard*, [0054].

The arrangement of components described above is symbolically illustrated in FIG. 1. An ‘ear canal microphone’ 10 is located within a user's ear canal 12, ***and a speaker 14 having a diaphragm 16 is arranged to direct sound into the user's ear canal in response to a driving signal 18.*** Ear canal microphone 10 produces an output signal 20 which varies with the instantaneous pressure in the ear canal. A controller 22 receives this signal and produces driving signal 18 so as to control the relationship between the instantaneous pressure in the ear canal and the speaker diaphragm's velocity such that the velocity is proportional to the instantaneous pressure.

*Beard*, [0026].

313. *Beard* further discloses speaker 14 receives the output of interface 50, which would be understood to include the output of the DSP. For example, as illustrated in Figure 3, the output of interface 50 is processed by a digital-to-analog converter (“DAC”), which is consistent with an output of a digital signal processor.

***An output 58 is preferably coupled to speaker 14 via a digital-to-analog converter (DAC) 59, the analog output of which is fed to the non-inverting input of op amp 36; additional outputs 60 are provided as needed. Some of the interface unit's inputs and outputs are***

preferably wireless connections, which are depicted as wavy lines in FIG. 3.

*Beard*, [0039].

Interface unit 50 can also include a digital audio memory for storing on board audio information like music. ***The interface can include digital processing power that allows actively modifying, storing, and sending digital audio information in various forms.*** With these features, the present system can operate in a wide range of modes selectable by the user. For instance, assuming the ear canal is substantially sealed by the system, if interface unit 50 sends no data via output 58 to DAC 59, the user will experience silence. That is, the components mounted in the ear will effectively block outside sound, controller 22 will emulate the acoustics of the user's open ear canal, and the zero input applied at non-inverting input 41 will further attenuate any outside sound leaking into the ear below the critical frequency.

314. *Beard* discloses that, among the sounds sent to speaker 14 are “swept sound signals” from the “DSP” of “interface 50.” *Beard*, [0044].

In practice, chirps, pseudorandom noise or ***swept sound signals can be used to determine the impulse transfer functions of the system.*** A feedforward noise cancellation scheme of this sort works effectively at lower frequencies, and since it is a feed forward system, there are no stability issues. ***Schemes of this type are typically implemented using one or more digital signal processors (DSPs).*** Methods for

implementing filters based on system transfer functions as described herein are well known in the art.

*Beard*, [0044].

315. As I explain in connection with Ground 1, *Epley* discloses sending the test signal to a speaker of an earphone (see §IX.A.6.d(ii), [1.2.2]).

316. As I explain in §IX.D.1, a person of ordinary skill in the art would have been motivated to apply *Epley's* seal assessment teachings (including sending a test signal to a speaker) to *Beard's* personal hearing system.

317. Moreover, *Beard* discloses “the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user.”

318. For example, *Beard* discloses its earphone includes “outer ear microphone,” a “speaker 14,” and “snug-fitting” “ear mold 102” that “fits snugly in the concha of the ear” and “extends into ear canal 106 and seals at 107” to block leakage of outside sound. *Beard*, [0012], [0026], [0040], [0052]-[0054], Fig. 4.

In a preferred embodiment, ***the personal hearing control system is fitted to seal the user's ear canal, with the speaker and ear canal microphone located on the inner ear side of the seal. The system preferably also includes a microphone located on the outer ear side of the seal***, and a handheld interface unit having one or more inputs

suitable for connection to respective sources of audio including the outer ear microphone. The interface is arranged to produce an output that varies with the audio received at a selected one of its inputs, and to couple the output—preferably wirelessly—to the speaker.

*Beard*, [0012].

The arrangement of components described above is symbolically illustrated in FIG. 1. An ‘ear canal microphone’ 10 is located within a user's ear canal 12, and ***a speaker 14 having a diaphragm 16 is arranged to direct sound into the user's ear canal*** in response to a driving signal 18. Ear canal microphone 10 produces an output signal 20 which varies with the instantaneous pressure in the ear canal. A controller 22 receives this signal and produces driving signal 18 so as to control the relationship between the instantaneous pressure in the ear canal and the speaker diaphragm's velocity such that the velocity is proportional to the instantaneous pressure.

*Beard*, [0026].

In a preferred embodiment, the personal hearing control system also includes ***a microphone 62 suitable for placement on the outer ear side of the ear canal***. The output 64 of this microphone is preferably buffered with an amplifier 66, digitized with an ADC 68, and provided to an input 70 of interface unit 50. ***Microphone 62 is preferably a high quality, low noise microphone which picks up outside sound at the ear***. Interface unit 50 might also include a built-in microphone 72 which allows the user to speak closely into it, for

privacy or when in a noisy environment, the output of which is buffered, digitized and provided to another of the interface's inputs 74.

*Beard*, [0040].

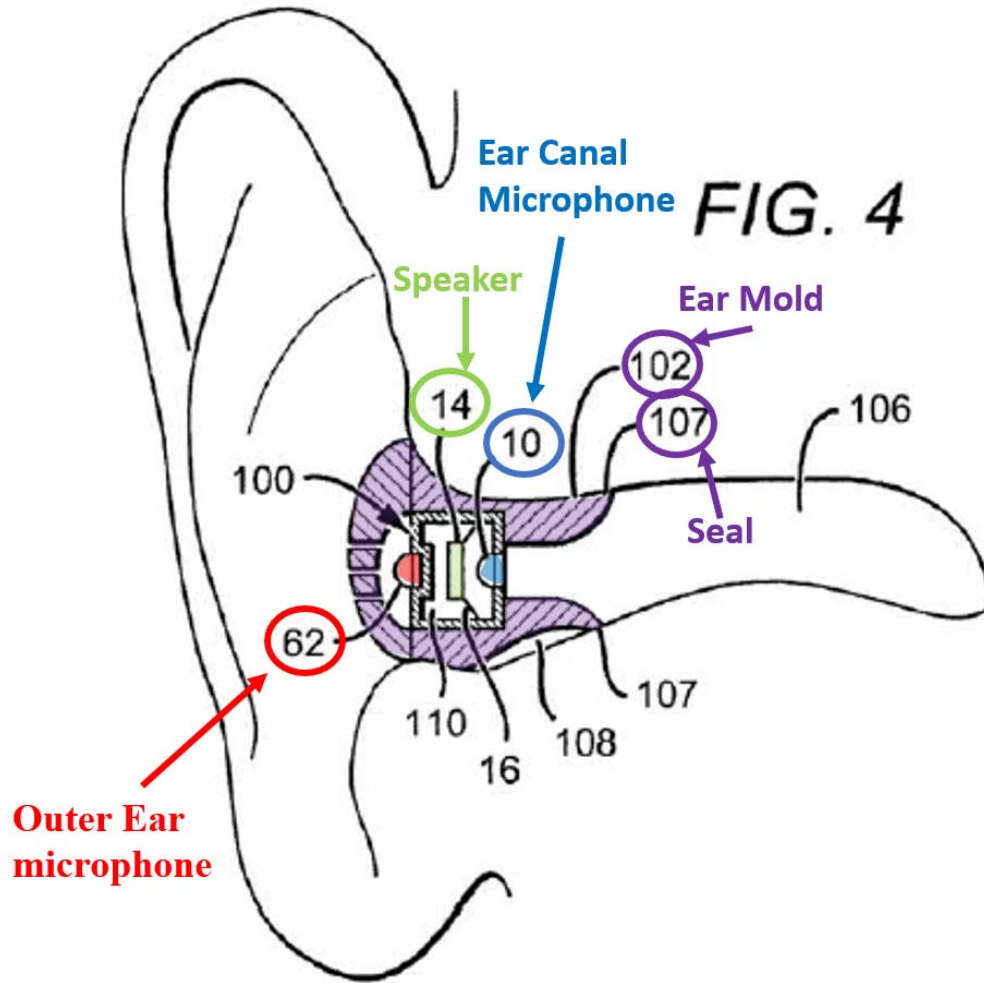
The present personal hearing control system is preferably fitted and calibrated for an individual user. In the preferred embodiment of the earphone element of the system, molds of the user's ears are made using the same technique used for fitting hearing aids. A soft silicon rubber casting of the user's ear canal, including the entire concha of the external pinna of the ear is made. This casting is then used to make a negative mold which is an accurate replica of the user's ear and ear canal. A soft molding compound such as Loctite Resonaid type 3596 is used to make a soft, ***snug-fitting ear mold with a cavity into which an electroacoustic unit containing speaker 14, ear canal microphone 10, and outer ear microphone 62, is mounted.*** The system components which directly interface with the components in the electroacoustic unit are preferably contained in a separate electronics unit (element 140 in FIG. 5, discussed below) located in a headband behind the head that also provides light medial pressure to the ear molds holding them in place. Due to their size and power requirements, the system's digital signal processing electronics are preferably housed within interface unit 50, rather than in the headband portion of the system.

***It is desirable that the ear mold element of the earphone effectively seal the ear canal, which serves to prevent higher frequency sounds from reaching the inner ear.*** However, a very small pressure

equalization vent should be provided to equalize the pressure between the sealed ear canal and the outside world, while substantially attenuating audio frequencies above  $F_{\min}$ . The back volume of the earphone unit which contains the microphones and speaker and acts as the back spring on the speaker diaphragm must also be vented to the outside world by a similar very small vent to provide pressure equalization.

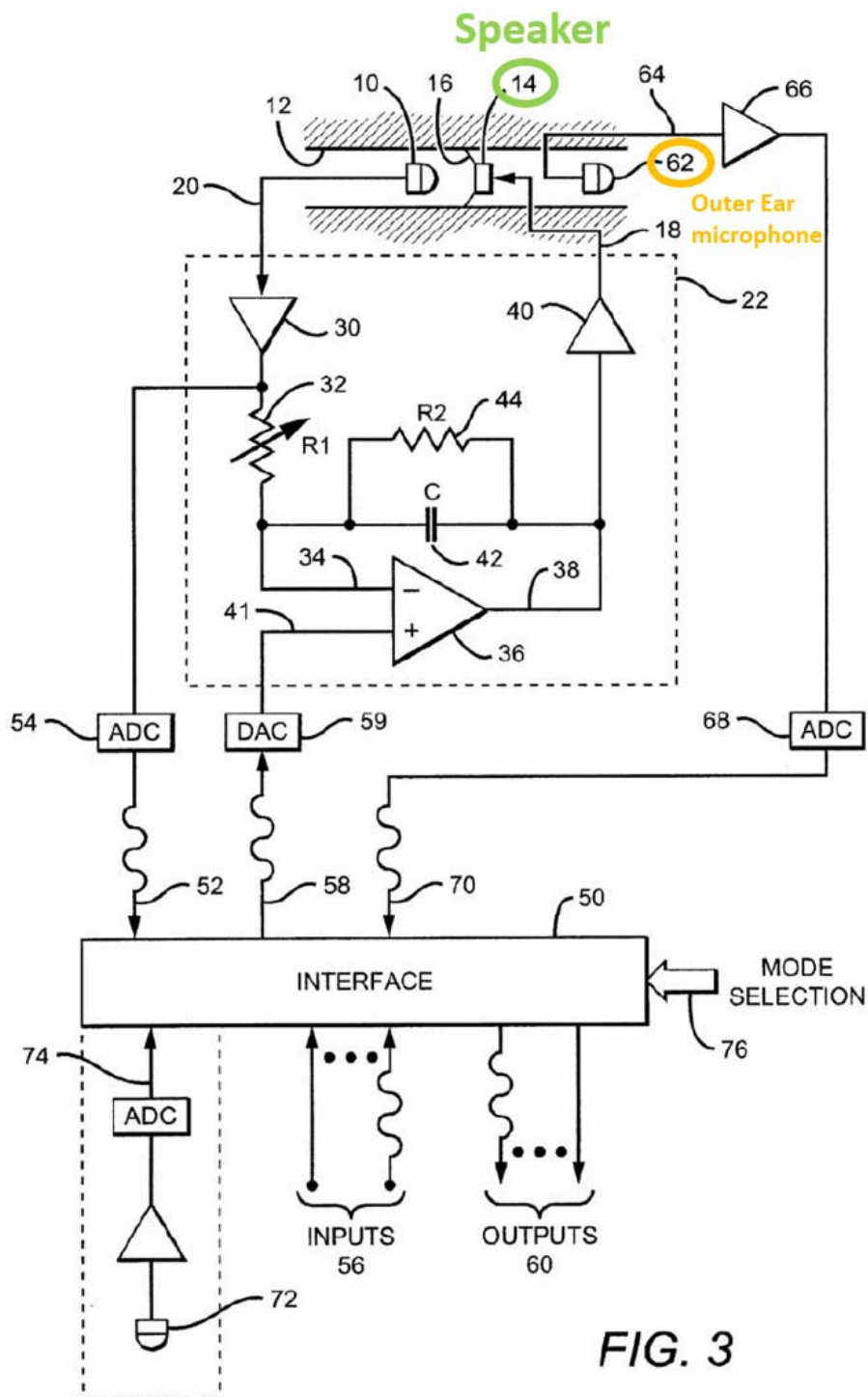
FIG. 4 is a cut-away drawing showing a preferred configuration for the ear-mounted elements of a system as described herein. Ear canal mic 10, ***outer ear mic 62 and speaker 14 of the earphone element are contained in a shell 100. The shell is mounted inside a custom ear mold 102 which fits snugly in the concha of the ear. Ear mold 102 extends into ear canal 106 and seals at 107. This seal blocks leakage of outside sound and provides for good low frequency response of the system.*** Seal 107 is preferably designed to leave an area 108 open to the atmosphere, to provide a relief path for pressure that might otherwise build due to mandible vibrations. The back seal volume 110 is preferably standardized, and is the determining contributor to the spring constant experienced by the diaphragm 16 of speaker 14. In practice, this volume should be less than 2 cubic centimeters. The molded earpiece can be designed to be held in place by the external ear itself, or can be held in place by a slight inward pressure by the headband or a spectacle configuration which combines the hearing devices with glasses.

*Beard*, [0052]-[0054].



Beard, Fig. 4 (annotated).

319. The connection of these components is shown in Figure 3.



*Beard*, Fig. 3.

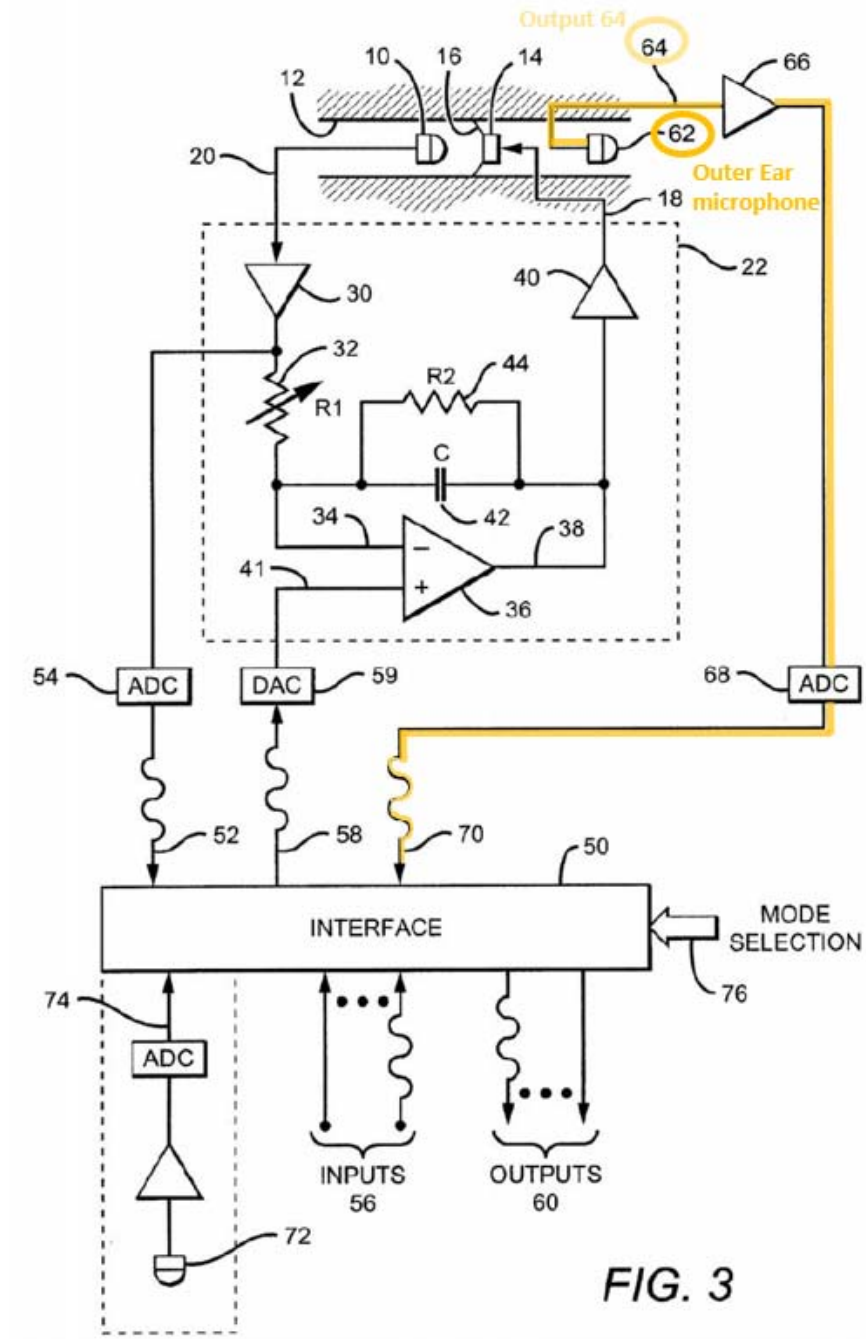
320. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Beard* in view of *Epley* to disclose or suggest that the DSP executes instructions stored in memory to send a known sound to **speaker 14** of the earphone (“sending the test signal to a speaker of an earphone”). Further, in my opinion, a person having ordinary skill in the art would have understood *Beard* to disclose or suggest that the earphone includes and **outer ear microphone 62** (“first microphone”), **speaker 14** (“speaker”), and **ear mold 102 that seals at 107** (“sealing section”). Finally, in my opinion, a person having ordinary skill in the art would have understood *Beard* to disclose or suggest that *Beard*’s earphone includes an **ear mold that seals at 107** to block leakage of outside sound into the ear canal (“wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user”).

**(iii) [1.2.4]: “receiving a first microphone signal from the first microphone;”**

321. In my opinion, *Beard* discloses or suggests “receiving a first microphone signal from the first microphone.”

322. As I explained in [1.2], *Beard* discloses “schemes” implemented by a “DSP.” Such instructions include instructions to receive the signal from the outer ear microphone 62. For example, *Beard* discloses a “noise cancellation scheme”

“implemented” using a “DSP[]” ([0044]) where “output 64” ([0040]) is received from “outer ear microphone 62.” ([0051]). This is illustrated in Figure 3 of *Beard*.



*Beard*, Fig. 3.

In a preferred embodiment, the personal hearing control system also includes ***a microphone 62 suitable for placement on the outer ear side of the ear canal. The output 64 of this microphone is preferably buffered with an amplifier 66, digitized with an ADC 68, and provided to an input 70 of interface unit 50.*** Microphone 62 is preferably a high quality, low noise microphone which picks up outside sound at the ear. Interface unit 50 might also include a built-in microphone 72 which allows the user to speak closely into it, for privacy or when in a noisy environment, the output of which is buffered, digitized and provided to another of the interface's inputs 74.

*Beard*, [0040].

In practice, chirps, pseudorandom noise or swept sound signals can be used to determine the impulse transfer functions of the system. ***A feedforward noise cancellation scheme of this sort*** works effectively at lower frequencies, and since it is a feed forward system, there are no stability issues. ***Schemes of this type are typically implemented using one or more digital signal processors (DSPs).*** Methods for implementing filters based on system transfer functions as described herein are well known in the art.

*Beard*, [0044].

A personal hearing control system in accordance with the present invention can allow the user to have normal hearing without removing the earphones. ***The audio received from outer ear microphone 62 may be equalized and the occlusion effect reduced or eliminated as***

***described above, with the result that outside sound is heard by the user as though the earphones are not there.*** This enables the system to act as a standard platform for 3D audio, as it provides for the standardization of the spatial audio signal characteristics received from outside ear microphone 62. In a preferred embodiment, the entire concha of the user's ear is filled such that the ear canal is substantially sealed, and outside microphone 62 is mounted flush on the outer surface of the system's earphone element so that the particular user's pinna and ear canal transfer function features are not present in the measured audio. In a 'normal' hearing (transparent) mode, the user's hearing system adapts to the audio heard exclusively by outside microphone 62, because all other sound leakage is suppressed below the level of audibility. The audio spatial characteristics of the signal from microphone 62 are associated by the user's internal auditory perception system with head turn and visual inputs so that the 3D audio cue set present in the audio signal from microphone 62 are adopted as the internally consistent set of 3D audio cues. The outside microphones are preferably mounted outside the filled concha with a standard baffle, so that they are acoustically consistent from user to user and are independent of individual pinna and ear canal acoustic characteristics that vary dramatically from individual to individual and create highly individualized head related transfer functions (HRTF's). The result is that the present system provides a universal platform to record and deliver standardized 3D audio signals that can be recorded and listened to by anyone.

*Beard*, [0051].

323. Moreover, to the extent the “processor is configured to execute the instructions to perform operations, the operations comprising” limitation is determined to be a means-plus-function limitation, *Beard*’s disclosure is the same or equivalent to the algorithm disclosed in the ’493 patent, which discloses “ASM 120 provides the measured ambient signal to processor.” ’493 patent, 9:3-4.

324. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Beard* to disclose or suggest that its processor execute schemes stored in the ROM to receive output 64 from the outer ear microphone 62 (receiving a first microphone signal from the first microphone).

**(iv) [1.2.5]: “calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination;”**

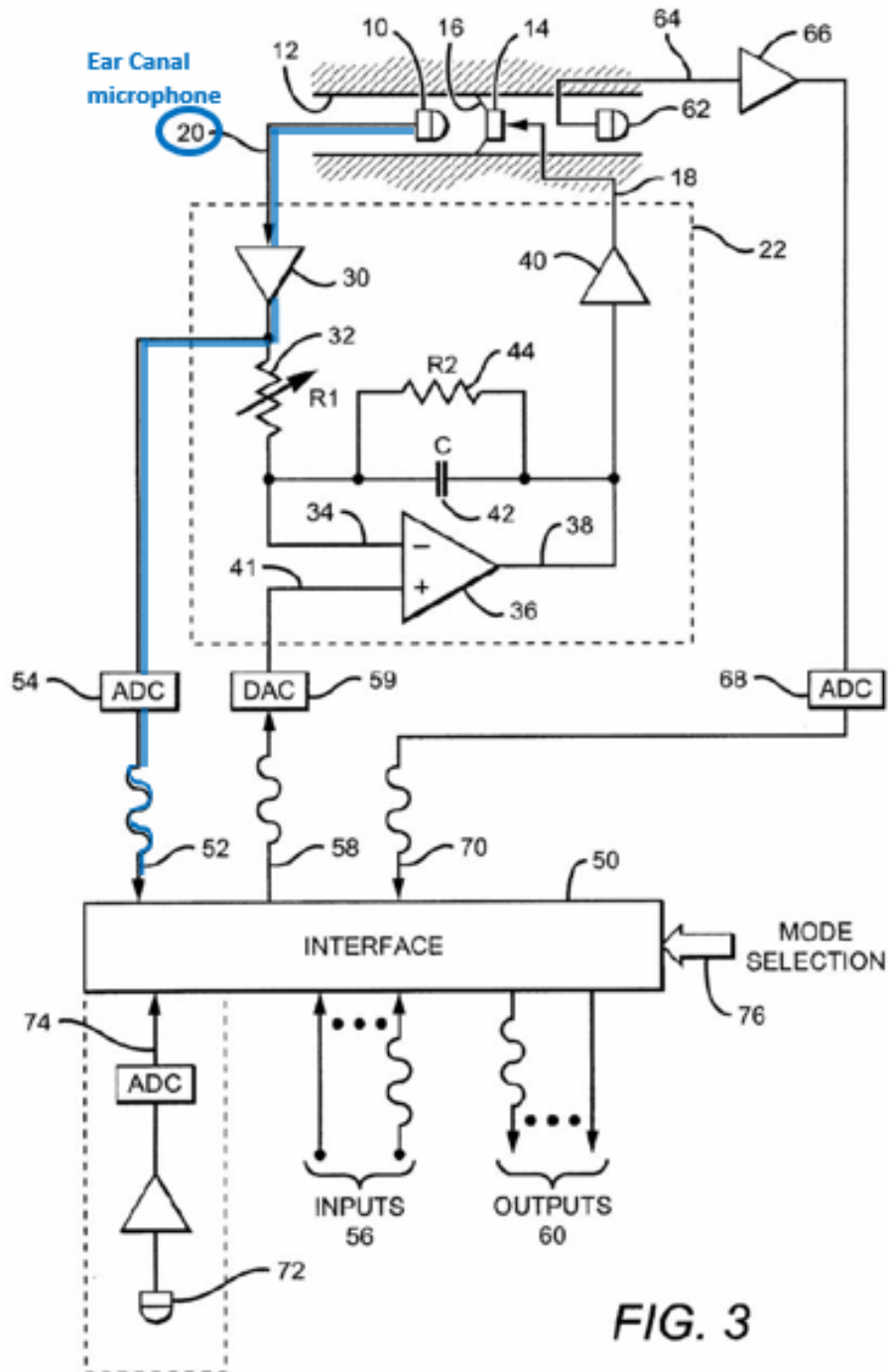
325. In my opinion, *Beard* in view of *Epley* and *Padhi* discloses or suggests “calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination.”

326. This limitation requires “calculating a value,” where the calculation uses “a first signal and a second signal.” I note that claim 1 does not specify what the recited “first signal” and “second signal” are or what generated them.

Regardless, *Beard* discloses a “first signal,” and *Epley* in view of *Padhi* discloses calculating a “value” using a “first signal” and a “second signal.”

327. As I explain above in connection with element [1.2], *Beard* discloses “schemes” implemented by a “DSP.”

328. *Beard* discloses DSP of “interface 50” receiving an “output signal 20” from “ear canal microphone 10” via “input 52,” as illustrated in Figure 3. *Beard*, [0011], [0026].



Beard, Fig. 3.

The present system includes *a microphone suitable for placement within a user's ear canal (the 'ear canal microphone') which*

*produces an output signal that varies with the instantaneous pressure in the ear canal*, a small loudspeaker which includes a diaphragm that directs sound into the ear canal, and a controller which receives the output signal from the ear canal microphone and provides a driving signal to the speaker.

Beard, [0011].

The arrangement of components described above is symbolically illustrated in FIG. 1. *An 'ear canal microphone' 10 is located within a user's ear canal 12*, and a speaker 14 having a diaphragm 16 is arranged to direct sound into the user's ear canal in response to a driving signal 18. *Ear canal microphone 10 produces an output signal 20 which varies with the instantaneous pressure in the ear canal*. A controller 22 receives this signal and produces driving signal 18 so as to control the relationship between the instantaneous pressure in the ear canal and the speaker diaphragm's velocity such that the velocity is proportional to the instantaneous pressure. Thus, when the pressure measured by microphone 10 increases, a force is applied to speaker 14 causing the velocity of diaphragm 16 to increase to the right in FIG. 1. When the relationship between diaphragm velocity and instantaneous air pressure is properly established over the range of sound frequencies that would normally be affected by the occlusion effect, the system emulates the acoustics of the user's open ear canal, thereby reducing or eliminating the occlusion effect that would otherwise be experienced.

Beard, [0026].

Further attenuation of outside sound can be achieved by using the signal from outer ear microphone 62 to produce a signal which cancels the audio leaking into the sealed ear canal. ***This is accomplished by providing one or more filters within interface 50*** which process the audio signal produced by outer ear microphone 62 such that a signal is coupled to speaker 14 that cancels sound that leaks from the outer ear to the inner ear side of the seal. ***Such a filter is preferably implemented by first determining the system transfer function between the audio signal produced by outer ear microphone 62 and the output of ear canal microphone 10, and the system transfer function between speaker 14 and the output of ear canal microphone 10.*** Once known, one or more filters can be implemented based on the transfer functions, which allow for the accurate cancellation of outside sound leaking into the sealed ear canal, as well as accurate high fidelity reproduction by the speaker of sound received by the outer ear microphone. The filters can be, for example, feedforward finite impulse response (FIR) digital filters, with each filter's coefficients calculated based on the system transfer functions.

*Beard*, [0043].

329. As I explain in connection with element [1.2.5] in Ground 1, in §IX.A.6.d(iv), *Epley's* seal assessment (using *Padhi's* cross-correlation) calculates a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-

correlation, a difference, or a combination. And as I explain in §IX.D.1IX.D.1, a person of ordinary skill in the art would have been motivated to apply the teachings of *Epley*'s seal assessment (using *Voix*'s threshold and *Padhi*'s cross-correlation) to *Beard*'s personal hearing system. In doing so, a person of ordinary skill in the art would have used a test signal (as taught by *Epley*) and *Beard*'s "output signal 20" received from "ear canal microphone 10" to perform the cross-correlation calculation.

330. Accordingly, in my opinion, a person having ordinary skill in the art would have understood that *Beard* in view of *Epley* and *Padhi* to disclose or suggest that instructions executed in the processor are configured to execute instructions stored in the memory to calculate a value that is the result of a non-phase cross correlation calculation ( $\alpha$ ) with the input signal from the ear canal microphone 10 and the known sound ("calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination").

- (v) **[1.2.6]-[1.2.7]: "comparing the value to a threshold to determine a seal quality between the first side and the second side;" and "generating a message to convey the seal quality; and sending the message to the user."**

331. In my opinion, *Beard* in view of *Epley*, *Padhi* and *Voix* teaches or suggests “comparing the value to a threshold to determine a seal quality between the first side and the second side.”

332. As I explain above in §IX.D.1, a person of ordinary skill in the art would have been motivated to apply the teachings of *Epley*’s seal assessment (using *Voix*’s threshold and *Padhi*’s cross-correlation) to *Beard*’s personal hearing system, and in doing so, *Beard*’s electronics module would send a warning to a display of its “computer.”

333. Accordingly, in my opinion, a person having ordinary skill in the art would have understood that *Beard* in view of *Epley*, *Padhi* and *Voix* discloses or suggests that instructions executed in a processor and stored in a memory include comparing a calculated value ( $\alpha$ ) to a threshold and determining the seal quality (i.e., *Padhi*’s score) between the two sides of the ear terminal (“comparing the value to a threshold to determine a seal quality between the first side and the second side”). Moreover, in my opinion, a person having ordinary skill in the art would have understood that *Beard* in view of *Epley* discloses or suggests that instructions executed by the DSP and stored in the memory include generating and sending a message to the user that conveys seal quality to the user (“generating a message to convey the seal quality” and “sending the message to the user”).

**3. Claim 4: “The audio device according to claim 1, wherein the speaker plays the test signal from the first side.”**

334. In my opinion, *Beard* in view of *Epley, Padhi* and *Voix* discloses or suggests “[t]he audio device according to claim 1, wherein the speaker plays the test signal from the first side.”

335. As I explain above in connection with claim 1, *Beard* in view of *Epley, Padhi* and *Voix* discloses the “audio device according to claim 1,” and in connection with elements [1.2.2] and [1.2.3], I explain outside the ear canal is the “first side.”

336. Moreover, *Beard* discloses a “speaker” that plays “driving signal 18” from the “inner ear side of the seal.” *Beard*, [0012].

In a preferred embodiment, the personal hearing control system is fitted to seal the user's ear canal, ***with the speaker and ear canal microphone located on the inner ear side of the seal.*** The system preferably also includes a microphone located on the outer ear side of the seal, and a handheld interface unit having one or more inputs suitable for connection to respective sources of audio including the outer ear microphone. The interface is arranged to produce an output that varies with the audio received at a selected one of its inputs, and to couple the output—preferably wirelessly—to the speaker.

*Beard*, [0012]. I explain this in more detail above in connection with element [1.2.2] to illustrate the speaker plays the driving signal from the first side, as illustrated in Figures 3 and 4.

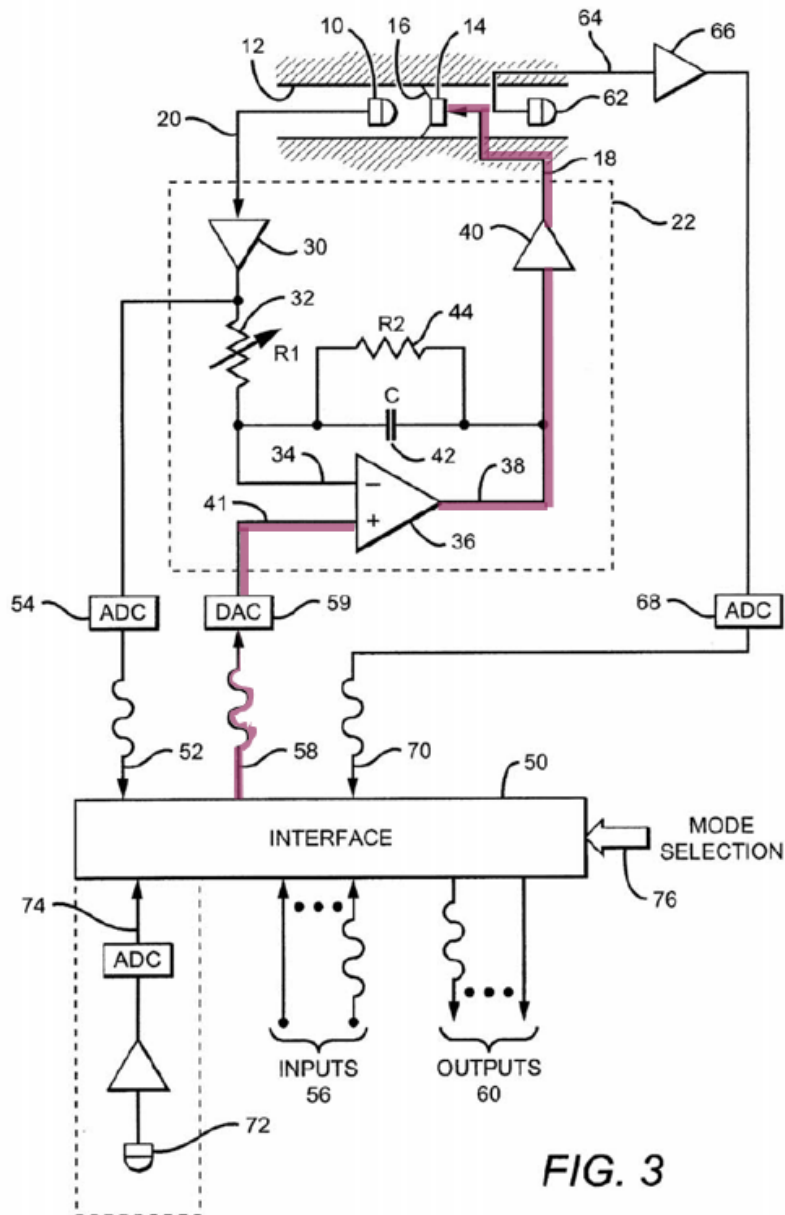
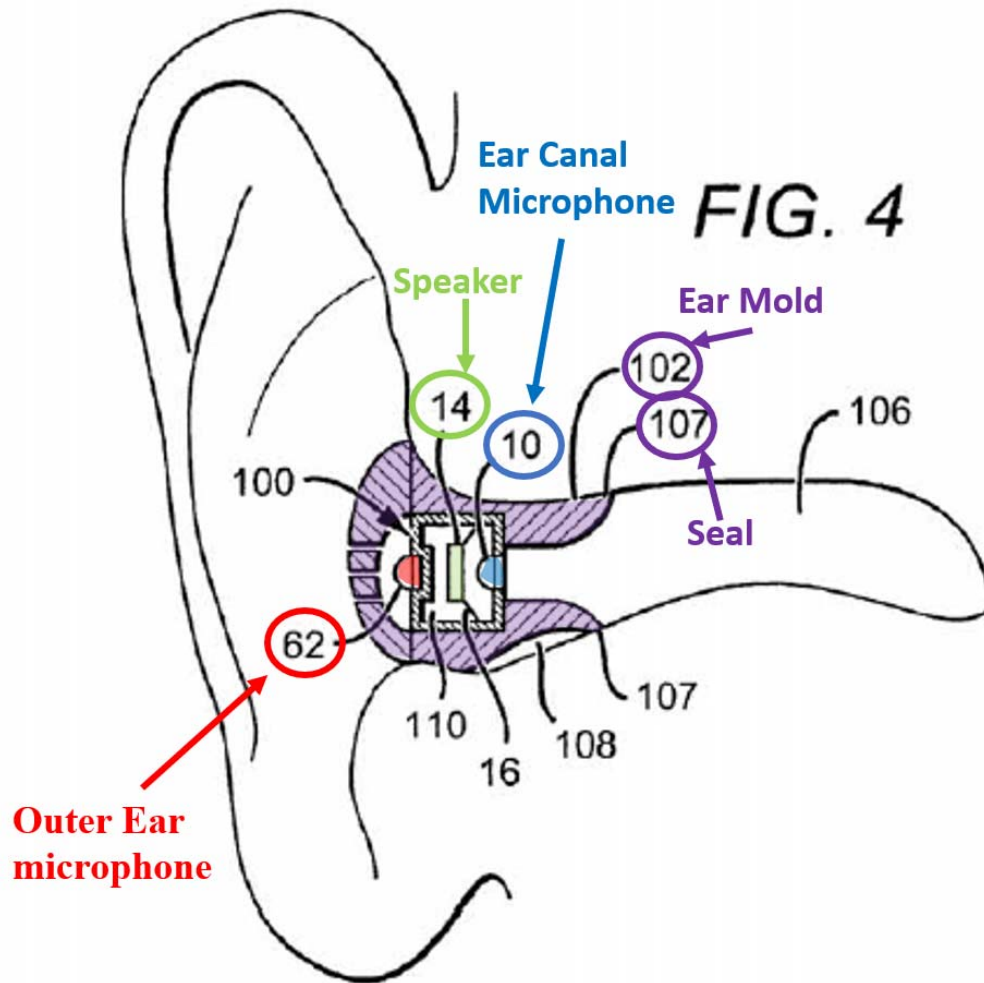


FIG. 3

*Beard*, Fig. 3 (annotated).



*Beard*, Fig. 4 (annotated).

337. As I explain in §IX.D.1, a person of ordinary skill in the art would have been motivated to apply the teachings of *Epley*'s seal assessment (including its test signal) to *Beard*'s personal hearing system.

338. Accordingly, in my opinion, a person having ordinary skill in the art would have understood that *Bear* in view of *Epley*, *Padhi* and *Voix* discloses or suggests the audio device of claim 1, wherein speaker 14 play a known sound from

the inside of the ear canal (“[t]he audio device according to claim 1, wherein the speaker plays the test signal from the first side.”).

**4. Claim 6: “The audio device according to claim 4, wherein the first microphone measures sound from the second side.”**

339. In my opinion, *Beard* in view of *Epley, Padhi* and *Voix* discloses or suggests “[t]he audio device according to claim 4, wherein the first microphone measures sound from the second side.”

340. As I explain above in connection with claim 4, *Beard* in view of *Epley, Padhi* and *Voix* discloses the “audio device according to claim 4.”

341. Moreover, *Beard* discloses microphone 62 “picks up outside sound at the ear.” *Beard*, [0040].

In a preferred embodiment, the personal hearing control system also includes ***a microphone 62 suitable for placement on the outer ear side of the ear canal.*** The output 64 of this microphone is preferably buffered with an amplifier 66, digitized with an ADC 68, and provided to an input 70 of interface unit 50. ***Microphone 62 is preferably a high quality, low noise microphone which picks up outside sound at the ear.*** Interface unit 50 might also include a built-in microphone 72 which allows the user to speak closely into it, for privacy or when in a noisy environment, the output of which is buffered, digitized and provided to another of the interface's inputs 74.

*Beard*, [0040]. *Beard* further refers to microphone 62 as “outer ear microphone 62.” *Beard*, [0043].

Further attenuation of outside sound can be achieved by using ***the signal from outer ear microphone 62*** to produce a signal which cancels the audio leaking into the sealed ear canal. This is accomplished by providing one or more filters within interface 50 which process the audio signal produced by ***outer ear microphone 62*** such that a signal is coupled to speaker 14 that cancels sound that leaks from the outer ear to the inner ear side of the seal. Such a filter is preferably implemented by first determining the system transfer function between the audio signal produced by ***outer ear microphone 62*** and the output of ear canal microphone 10, and the system transfer function between speaker 14 and the output of ear canal microphone 10.

*Beard*, [0043].

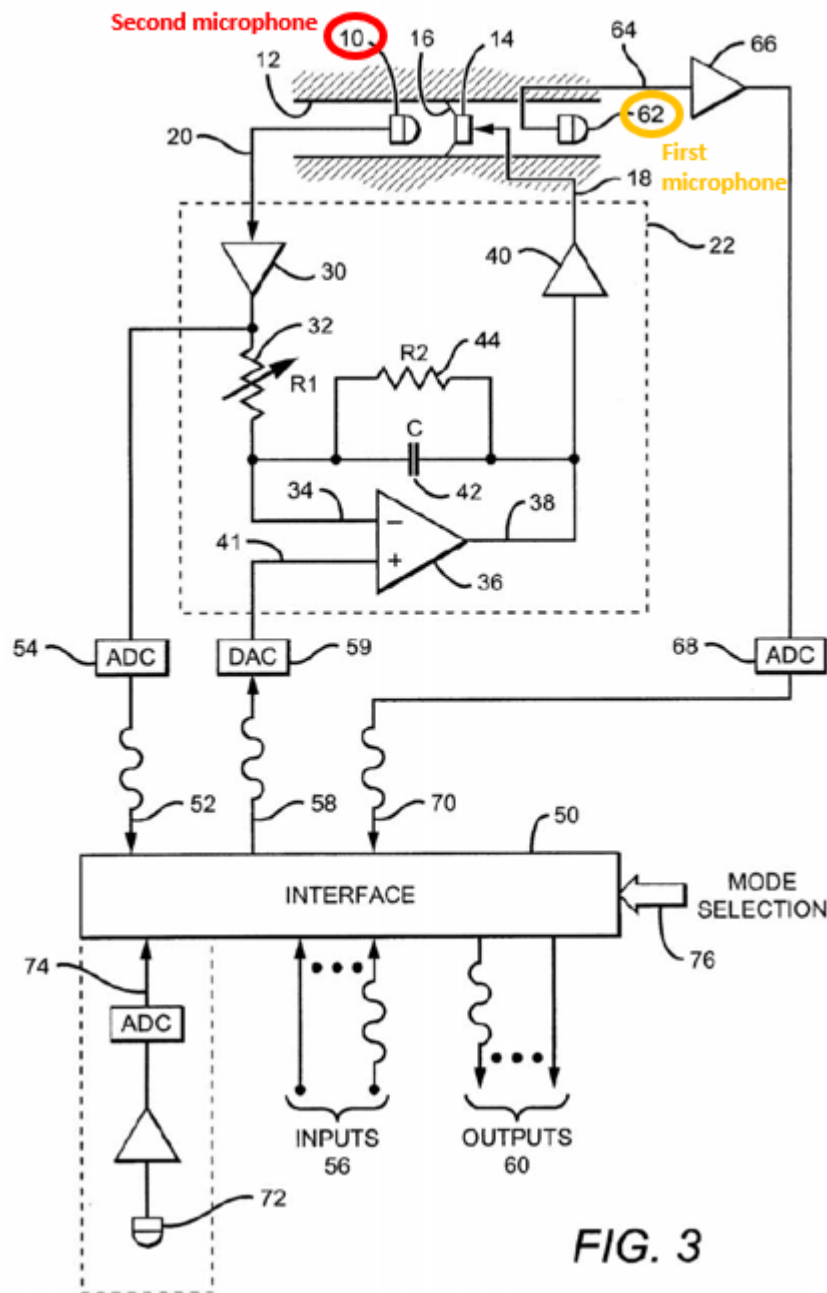
342. Accordingly, in my opinion, a person having ordinary skill in the art would have understood that *Bear* in view of *Epley*, *Padhi* and *Voix* discloses or suggests the audio device of claim 4, wherein outer ear microphone 62 measures sound from outside the ear (“[t]he audio device according to claim 4, wherein the first microphone measures sound from the second side”).

**5. Claim 13: “The audio device according to claim 1, wherein the earphone includes a second microphone.”**

343. In my opinion, *Beard* in view of *Epley, Padhi* and *Voix* discloses or suggests “[t]he audio device according to claim 1, wherein the earphone includes a second microphone.”

344. As I explain above in connection with claim 1, *Beard* in view of *Epley, Padhi* and *Voix* discloses the “audio device according to claim 1.”

345. *Beard* further discloses “[a]n ‘ear canal microphone’ 10 is located within a user's ear canal 12....” *Beard*, [0012]. The ear canal microphone 10 is illustrated in Figure 3.



*Beard*, Fig. 3 (annotated).

346. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Beard* in view of *Epley*, *Padhi* and *Voix* to disclose or

suggest an earphone that includes an ear canal microphone 10 (“wherein the earphone includes a second microphone”).

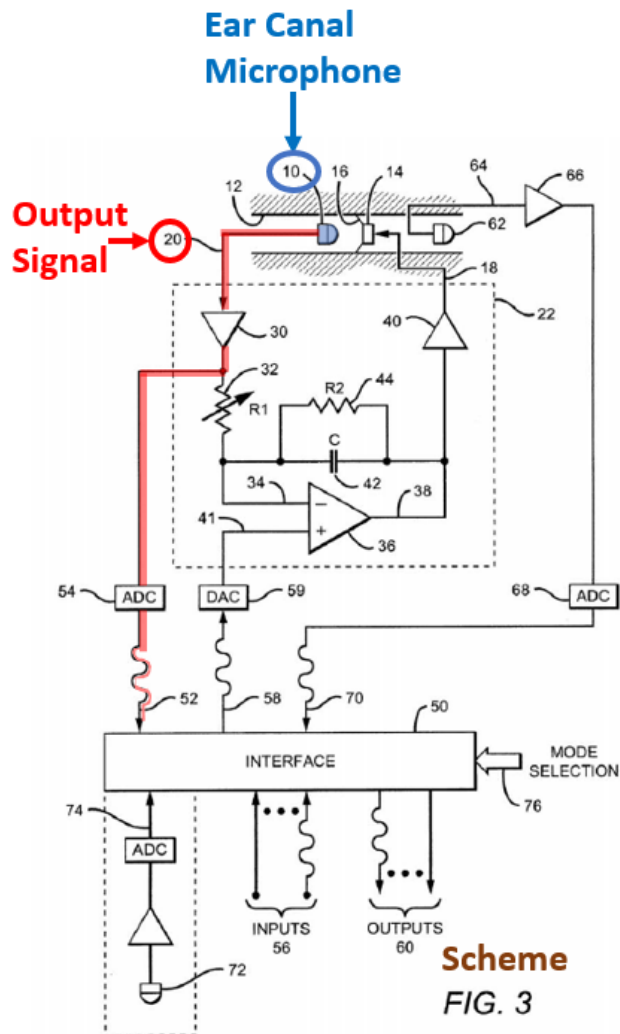
**6. Claim 14: “The audio device according to claim 13, wherein the operations further comprise: receiving a second microphone signal from the second microphone.”**

347. In my opinion, *Beard* in view of *Epley, Padhi* and *Voix* discloses or suggests “[t]he audio device according to claim 13, wherein the operations further comprise: receiving a second microphone signal from the second microphone.”

348. As I explain above in connection with claim 13, *Beard* in view of *Epley, Padhi* and *Voix* discloses the “audio device according to claim 13.”

349. As I explained in [1.2], *Beard* discloses “schemes” implemented by a “DSP.” Such instructions include instructions to receive the signal from the ear canal microphone 10.

350. For example, as I explain in connection with element [1.2.5], *Beard* discloses DSP of “interface 50” receiving an “output signal 20” from “ear canal microphone 10” via “input 52,” as illustrated in Figure 3.



*Beard*, Fig. 3

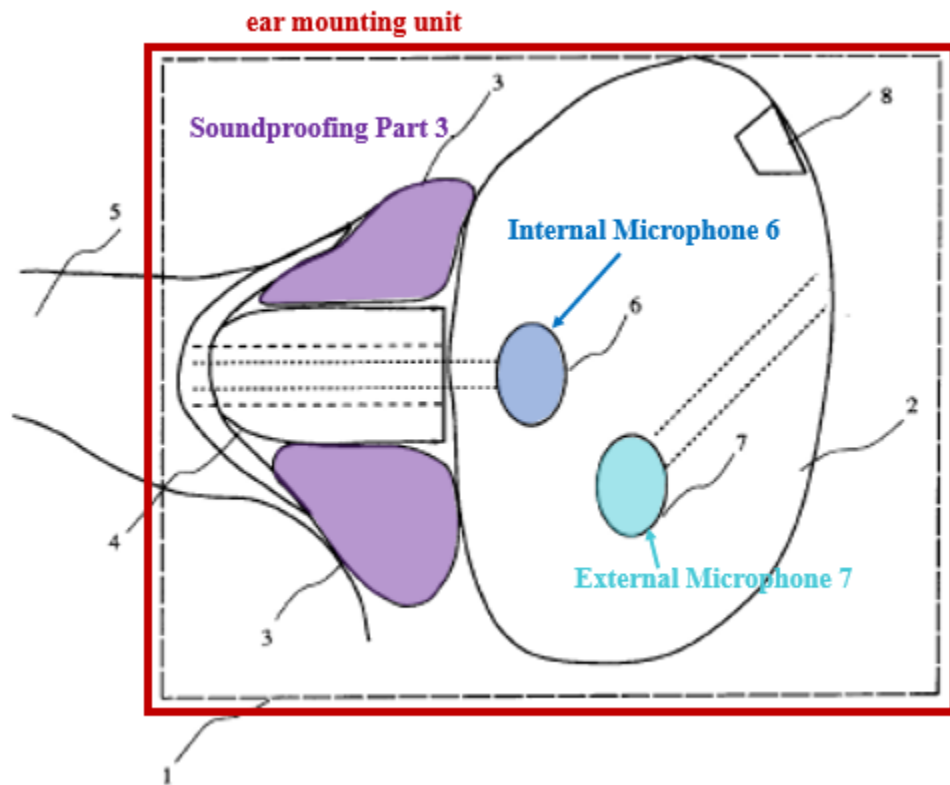
351. Moreover, to the extent the “processor is configured to execute the instructions to perform operations, the operations comprising” limitation is determined to be a means-plus-function limitation, *Beard*’s disclosure is the same or equivalent to the algorithm disclosed in the ’493 patent, which discloses “ECM 106 is configured to measure the test signal in ear canal 124 and provide the measured test signal to processor 116.” ’493 patent, 8:58-60.

352. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Beard* in view of *Epley*, *Padhi* and *Voix* to disclose or suggest that DSP is configured to execute instructions stored in memory to receive a second microphone signal from the ear canal microphone 10 (“receiving a second microphone signal from the second microphone”).

**E. Ground 5: *Beard* in view of *Epley*, *Padhi*, *Voix* and *Hayashi*; claims 15 (“The audio device according to claim 14, wherein the first signal is the first microphone signal and the second signal is the second microphone signal.”), 16 (“The audio device of claim 14, wherein the first signal is a modified first microphone signal and the second signal is the second microphone signal.”) and 17 (“The audio device of claim 14, wherein the first signal is a modified first microphone signal and the second signal is a modified second microphone signal.”)**

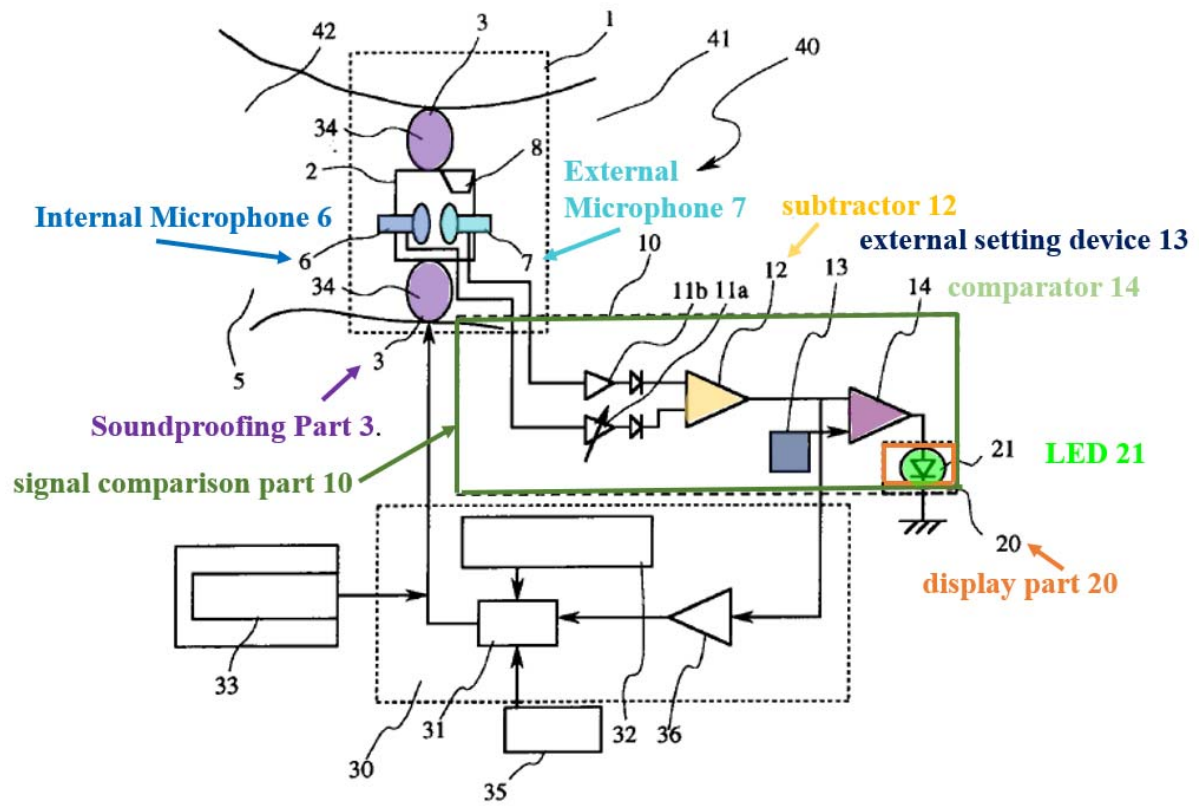
353. In my opinion, *Beard* in view of *Epley*, *Padhi*, *Voix* and *Hayashi* discloses or suggests claims 15-17.

354. *Hayashi* discloses determining the similarity between two microphone signals to evaluate the seal of an in-ear device. In Figure 2, *Hayashi* discloses an **ear mounting unit 1** with an **external microphone 7**, **internal microphone 6**, and a **soundproofing part 3**.



*Hayashi*, Fig. 2.

355. *Hayashi* discloses an algorithm for calculating a value with the internal and external microphone signals to evaluate the level of seal. For context, *Hayashi*'s Figure 1 is reproduced below with relevant components annotated.



Hayashi, Fig. 1.

356. Hayashi discloses that **signal comparison part 10** receives input signals from **internal microphone 6** and **external microphone 7** and supplies them to **subtractor 12**. Hayashi, [0013].

The **signal comparison part 10** is composed of an amplifier 11a having a gain adjustment function, an amplifier 11b, a **subtractor 12**, an **external setting device 13**, and a **comparator 14**.

Note that the amplifier 11a having a gain adjustment function is provided with a function that may change a gain such that a target soundproofing capability may be set from the outside. The **signal comparison part 10** receives as input a signal from the **internal**

**microphone 6** directed toward the space 42 of the **ear canal 5** and a signal from the **external microphone 7** directed toward the free space 41. *The signal from the external microphone 7 is amplified by the amplifier 11b and inputted to the subtractor 12. The signal from the internal microphone 6 is also input to the subtractor 12 after an amplification degree is adjusted by the amplifier 11a having a gain adjustment function.* The **subtractor 12** is a differential amplifier and amplifies a potential, that is, a sound pressure difference, obtained by subtracting an **input signal of the internal microphone** from an **input signal of the external microphone**. An output signal of the **subtractor 12** is compared to a reference voltage of the **external setting device 13** by the **comparator 14**. The **comparator 14** is a comparator, and outputs an output voltage for turning on an **LED 21** when an input voltage exceeds a comparison voltage, and the output voltage is 0 V when the input voltage is less than the comparison voltage. The output voltage of the **comparator 14** is applied as-is to the **LED 21** of the **display part 20**. Note that the **external setting device 13** is an operation part wherein the user sets a threshold corresponding to a predetermined sound pressure difference as the reference voltage.

Hayashi, [0013].

357. Hayashi's **subtractor 12** subtracts signals from the **internal microphone 6** and **external microphone 7**, and **comparator 14** compares the output of the subtractor to a reference voltage threshold set by **external setting**

**device 13**. The reference voltage threshold corresponds to a predetermined sound pressure difference. If the input to the **comparator 14** is less than the threshold, the comparator outputs 0 volts. Otherwise, the **comparator 14** outputs a voltage to light up **LED 21** in **display part 20** to indicate the level of sound leakage through the **soundproofing part 3**. *Id.*, [0013]-[0014]; *id.*, [0018].

The **signal comparison part 10** is composed of an amplifier 11a having a gain adjustment function, an amplifier 11b, a **subtractor 12**, an **external setting device 13**, and a **comparator 14**.

Note that the amplifier 11a having a gain adjustment function is provided with a function that may change a gain such that a target soundproofing capability may be set from the outside. The **signal comparison part 10** receives as input a signal from the **internal microphone 6** directed toward the space 42 of the **ear canal 5** and a signal from the **external microphone 7** directed toward the free space 41. The signal from the **external microphone 7** is amplified by the amplifier 11b and inputted to the **subtractor 12**. The signal from the **internal microphone 6** is also input to the **subtractor 12** after an amplification degree is adjusted by the amplifier 11a having a gain adjustment function. The **subtractor 12** is a differential amplifier and amplifies a potential, that is, a sound pressure difference, obtained by subtracting an **input signal of the internal microphone** from an **input signal of the external microphone**. An output signal of the **subtractor 12** is compared to a reference voltage of the **external setting device 13** by the **comparator 14**. The **comparator 14** is a

comparator, and outputs an output voltage for turning on an **LED 21** when an input voltage exceeds a comparison voltage, and the output voltage is 0 V when the input voltage is less than the comparison voltage. The output voltage of the **comparator 14** is applied as-is to the **LED 21** of the **display part 20**. Note that the **external setting device 13** is an operation part wherein the user sets a threshold corresponding to a predetermined sound pressure difference as the reference voltage.

*Hayashi, [0013].*

The **display part 20** is composed of the **LED 21** and the **LED 21** is turned on based on the input signal from the **signal comparison part 10**. Note that, in the present embodiment, display is performed using the **LED 21**, but a method may be used to notify the user of the leakage amount of the external sound by a sound, vibration, or the like.

*Hayashi, [0014].*

Next, a state when the device of the present embodiment is mounted on an ear will be described using FIG. 2. As illustrated in the descriptive diagram of the **ear mounting unit 1** of FIG. 2, the ear canal 5 is plugged by the earplug 4 and the **soundproofing part 3** to soundproof the external sound 40. When collecting the external sound 40, the **internal microphone 6** collects sound in the ear canal 5 and the **external microphone 7** collects sound of the free sound field 41 and these are converted into an electrical signal. A phase of the signal

is different depending on a disposition of the **internal microphone 6** and the **external microphone 7** in the housing part 2. Therefore, after determining in advance a disposition such that the phase difference is minimized, the housing part 2 is configured. Thus, it is possible to compare the same sound without being affected by the phase difference. The same external sound 40 collected by the **internal microphone** and the **external microphone** is converted into an electrical signal and subtracted by the **subtractor 12** of the **signal comparison part 10**. The subtracted signal is compared to the threshold set in the **external setting device 13** by the **comparator 14**. Then, the output signal of the **signal comparison part 10** becomes an input signal to the **LED 21** of the **display part 20**. For example, when configuring such that the threshold is set to a voltage corresponding to the predetermined sound pressure difference and the **LED 21** of the **display part 20** is lit up when the input signal is at or above the threshold, the user of the device of the present embodiment may know the leakage amount of the sound by looking at the lighting state of the **LED 21**. Furthermore, when the degree of sealing of the **soundproofing part 3** is evaluated, the external sound 40 or a presentation sound generated from an external sound generating speaker 8 is collected by the **internal microphone 6**. At this time, the sound collected by the **internal microphone 6** is a sound leaked into the ear canal 5. Therefore, the degree of sealing of the **soundproofing part 3** may be evaluated by changing and generating the presentation sound and monitoring the sound pressure of the leaked sound using a presentation sound pressure monitor (not illustrated).

*Hayashi*, [0018].

358. Thus, *Hayashi* discloses that two microphone signals measuring sound from opposite sides of a sealing section (**soundproofing part 3**) in an ear mounted unit may be utilized in a calculation to determine the seal quality.

359. In my opinion, in applying *Hayashi*'s teachings of using two microphone signals to *Beard* as modified by using *Epley*'s seal assessment, *Voix*'s threshold and *Padhi*'s cross-correlation, for limitation [1.2.4], *Beard*'s ear canal microphone 10 and outer ear microphone 62 would provide inputs to *Padhi*'s **cross correlation function 134** instead of *Epley*'s known sound and the signal from *Beard*'s ear canal microphone 10.

360. In my opinion, the analysis of limitations [1.2.5] and [1.2.6] remains the same, except that in my opinion a person having ordinary skill in the art would have recognized that where two microphone signals are used in the calculation, a lower cross correlation calculated value would correspond to a higher seal quality (that is, the less correlated the two signals are, the better the seal), and would adjust the threshold comparison step accordingly before generating and sending the message to the user.

361. In my opinion a person having ordinary skill in the art would have been motivated to perform the calculation with *Beard*'s ear canal microphone 10 and outer ear microphone 62 instead of performing the calculation with *Epley*'s

known sound and the signal from *Beard*'s ear canal microphone 10. *Hayashi* explains that, by using two microphone signals for a seal test, "it is possible to enhance a measuring accuracy of a pressure difference of sound collected from both microphones." *Hayashi* [0006]. And because *Beard* already teaches two microphones, application of *Hayashi*'s use of two microphone signals for a seal test would have been straightforward. Both microphone signals are provided to interface unit 50, and thus, would be available to the DSP performing a seal test in *Beard* as modified by the teachings of *Epley*, *Padhi*, *Voix* and *Hayashi*.

362. For the reasons I explain above in connection with claims 7, 8, 10 and 11 (see §§IX.A.10 - A.13) it would have been obvious to modify the input signals to the cross-correlation using the pre-processing described in *Padhi*, and that the resulting value obtained from the cross-correlation calculation would have been obtained from the original microphone signals because the resultant value  $\alpha$  was generated "using" the microphone signals. Moreover, for the reasons I explain above in connection with claims 7, 8, 10 and 11 (see §§IX.A.10 - A.13), to the extent it is determined that the claims require that "the first microphone signal" of claims 16 and 18 and "the second microphone signal" of claims 16 and 17 requires that the signals output from *Beard*'s ear canal microphone 10 and outer ear microphone 62 not be modified before entering the *Padhi*'s **cross correlation function 134**, a person having ordinary skill in the art would have understood that

it would have been a simple design choice to not first perform pre-processing on them in [section 110](#) to “modify” them.

## **X. SECONDARY CONSIDERATIONS**

363. I am not aware of any evidence in the '493 patent's prosecution history or elsewhere supporting any secondary considerations arguments, or evidence of nexus of such alleged evidence to the challenged claims. *See generally* EX1004. To the extent Patent Owner asserts the existence of any secondary considerations in its responses, I reserve the right to address any such evidence.

## **XI. CONCLUSION**

364. The findings and opinions set forth in this declaration are based on my work and examinations to date.

365. I may continue my examinations. I may also receive additional documentation and other factual evidence over the course of this IPR that will allow me to supplement and/or refine my opinions. I reserve the right to add to, alter, or delete my opinions and my declaration upon discovery of any additional information. I reserve the right to make such changes as may be deemed necessary.

366. In signing this declaration, I recognize that the declaration will be filed as evidence in an IPR before the PTAB. I also recognize that I may be

subject to cross-examination in the case and that cross-examination will take place within the United States. If cross-examination is required of me, I will appear for cross-examination within the United States during the time allotted for cross-examination.

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

Date: 7/1/2024

  
Chris Kyriakakis, Ph.D