

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

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

Petitioners

v.

STATON TECHIYA, LLC,

Patent Owner

Case IPR2024-01034

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

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

Samsung et al. v. Techiya IPR2024-01034 Exhibit 1002
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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).¹ My opinions are set forth below.

¹ 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. Patent No. 6,567,524 ("*Svean*") (Ex. 1007);
- U.S. Patent Application Publication No. 2005/0096899 ("*Padhi*") (Ex. 1008);

- Certified Translation of Japanese Patent Application Publication 2002-204500 (“*Hayashi*”) (Ex. 1009);
- Japanese Patent Application Publication 2002-204500 (Ex. 1010);
- U.S. Patent No. 5,182,774 (“*Bourk*”) (Ex. 1011);
- U.S. Patent No. 8,401,198 (“*Oh*”) (Ex. 1012);
- Chu, W. T., *A deterministic broad-band signal for acoustical measurements*, Inter-Noise 87 1199-1202 (“*Chu*”) (Ex. 1013);
- U.S. Patent No. 7,158,643 (“*Lavoie*”) (Ex. 1014);
- U.S. Patent Application Publication No. 2005/0157891 (“*Johansen*”) (Ex. 1015);
- WIPO Publication No. WO2001089083A1 (“*Anagnos*”) (Ex. 1016);
- ’836 patent *Ex Parte Reexamination*, Final Rejection (07/28/2023) (Ex. 1017);
- U.S. Patent No. 6,048,320 (“*Brainard*”) (Ex. 1018);
- Schneider, T. et al., *A Dual-Channel MLS-Based Test System for Hearing-Aid Characterization*, J. Audio Eng. Soc., Vol. 41, No. 7/8 (1993) (Ex. 1019);
- U.S. Patent No. 6,687,377 (“*Voix*”) (Ex. 1020);
- Claim Construction Order from the district court where this patent is being asserted (Ex. 1021);

- '836 patent *Ex Parte Reexamination*, Appeal Brief of Staton Techiya LLC (11/21/2023) (Ex. 1023); and

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: *Svean, Voix, and Padhi* renders obvious claims 1, 2, 4-8, 10, 11, 13, 14, and 18-20.
- Ground 2: *Svean, Voix, Padhi, and Hayashi* render obvious claims 3 and 15-17.
- Ground 3: *Svean, Voix, Padhi, and Bourk* render obvious claims 9 and 12.

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.

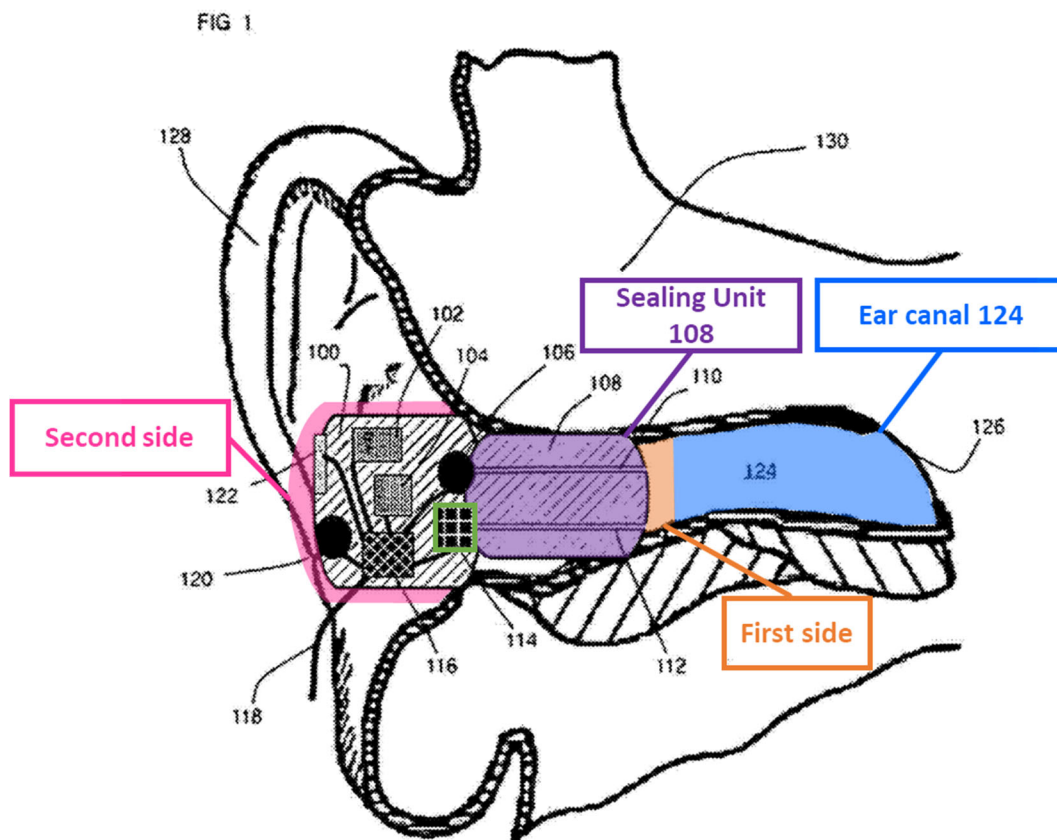
40. To remedy the purported problems in the art, the '493 patent proposes:

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.

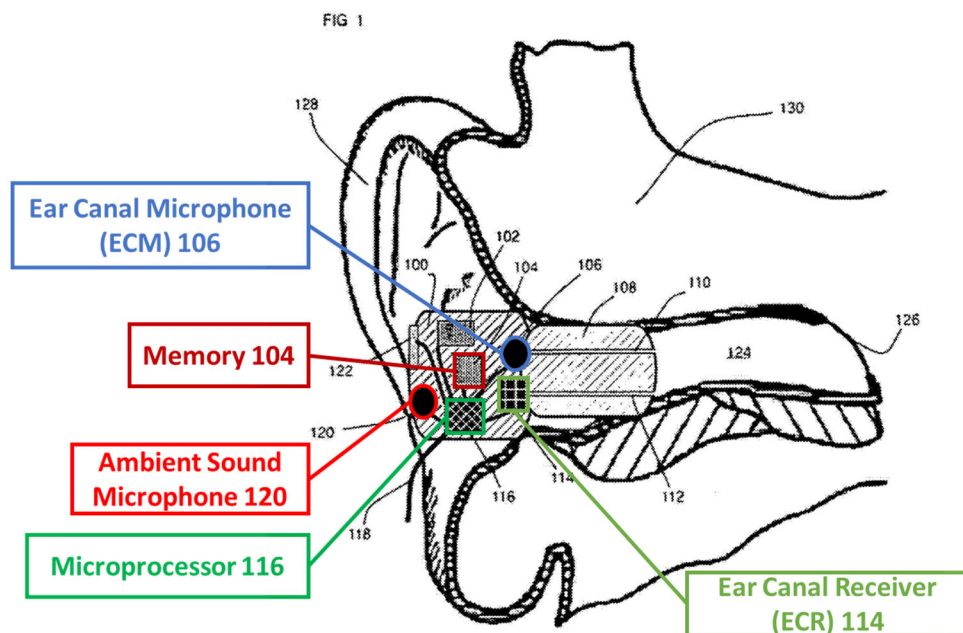
41. Figure 1, reproduced and annotated below, shows an earpiece comprising **sealing unit 108** “having a **first side** corresponding to **ear canal 124** and a **second side** corresponding to the ambient environment.”² *Id.*, 4:34-36.



Svean, Figure 1.

² All emphases are added in this Declaration unless otherwise noted.

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 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 14.³ *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**.

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

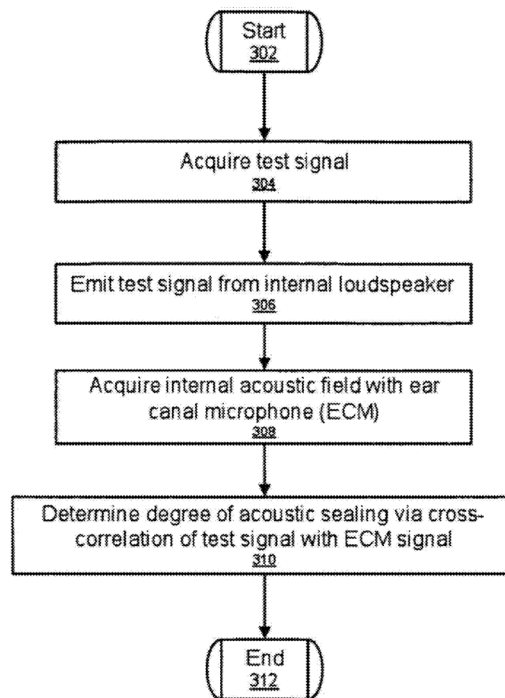


FIG. 3

'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.

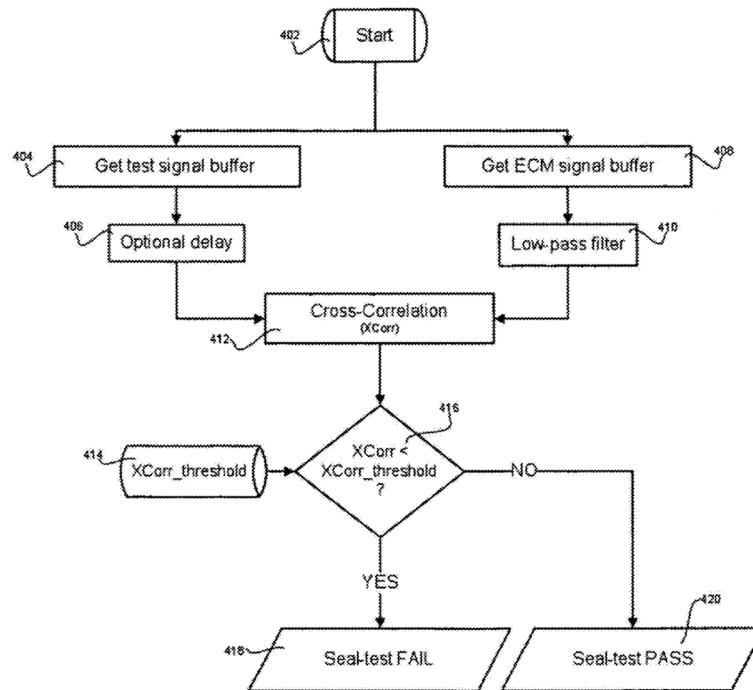


FIG. 4

'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

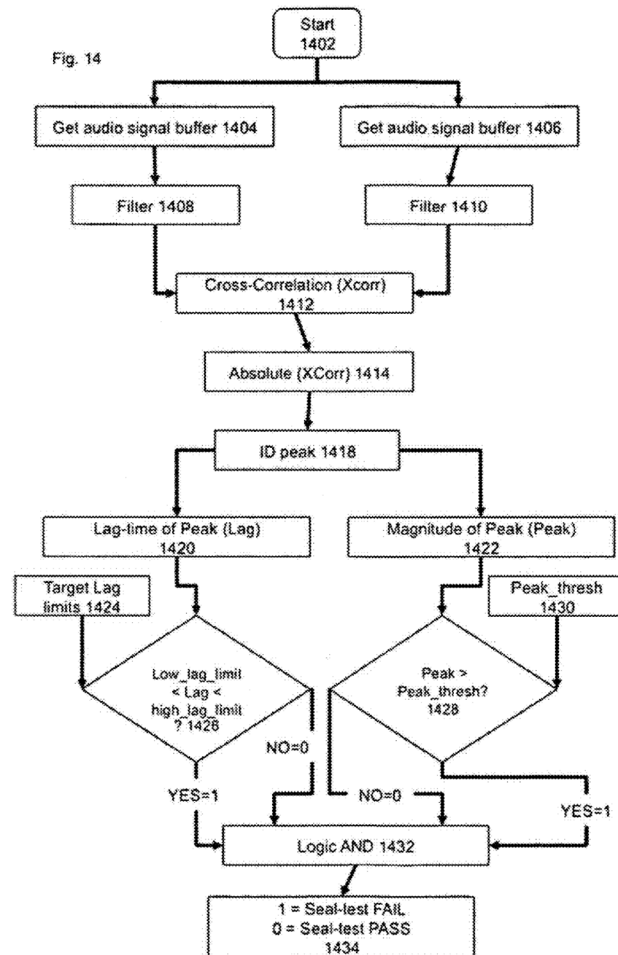
buffer is optionally delayed by digital delay unit 406. The function of delay unit 406 is to time-align the emitted test signal with the ECM signal so the cross-correlation is sensitive to changes in ear seal.

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, the Examiner noted that, regarding independent claim 1, *Svean* was the “closest prior art of record” to independent claim 1 and disclosed several limitations. *Id.*, 135-36. However, the Examiner stated that *Svean* “fails to teach the combination of an audio device configured to test a seal of an earphone comprising: a memory that stores instructions; and a processor, wherein the processor is configured to execute the instructions to perform operations, the operations comprising: generating a test signal, wherein the test signal includes multiple frequency components, wherein at least one component lies between a range of 30Hz and 200Hz; sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user, receiving a first microphone signal from the first microphone; calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination; comparing the value to a threshold to determine a seal quality between the first side and the second side; generating a message to convey the seal quality; and sending the message to the user.” *Id.*, 136 (Examiner’s underlining).

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.

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

Claim 3	
3	The audio device according to claim 1, wherein the message is a visual message sent to a display of the audio device.
Claim 4	
4	The audio device according to claim 1, wherein the speaker plays the test signal from the first side.
Claim 5	
5	The audio device according to claim 4, wherein the first microphone measures sound from the first side.
Claim 6	
6	The audio device according to claim 4, wherein the first microphone measures sound from the second side.
Claim 7	
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.
Claim 8	
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.
Claim 9	
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.
Claim 10	
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.
Claim 11	
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.

Claim 12	
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.
Claim 13	
13	The audio device according to claim 1, wherein the earphone includes a second microphone.
Claim 14	
14	The audio device according to claim 13, wherein the operations further comprise: receiving a second microphone signal from the second microphone.
Claim 15	
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.
Claim 16	
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.
Claim 17	
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.
Claim 18	
18	The audio device of claim 1, wherein the seal quality is good if the value is at or above the threshold value.
Claim 19	
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.

Claim 20	
20	The audio device of claim 1, wherein the sealing section is an eartip.

VIII. OVERVIEW OF THE PRIMARY PRIOR ART REFERENCES

60. In my opinion, and as I elaborate on below, the prior art references described in this section are within the same field of endeavor of the '493 patent and/or are reasonably pertinent to problems that the '493 patent seeks to address.

61. Like the '493 patent, *Svean*, *Voix*, and *Hayashi* similarly disclose embodiments for analyzing the acoustic seal of an ear worn device. *See, e.g., Svean*, 11:61-13:38; *Voix*, 3:33-58, 4:49-6:22, 6:53-7:48, 8:56-10:20; *Hayashi*, [0018]. *Padhi* is directed to techniques for comparing the similarity of two audio signals (*Padhi*, [0001], [0004]-[0006]), which is pertinent to the acoustic seal analysis problem addressed by the '493 patent because—as references such as *Svean*, *Voix*, and *Hayashi* disclose and as was well known to a person having ordinary skill in the art—assessing acoustic seal involves a determination of the similarity of audio signals. Further, like the '493 patent, *Bourk* is directed to a device that provides audio to a user. *Bourk*, Abstract.

A. *Svean*

62. *Svean* is titled “Noise Protection Verification Device.” *Svean*, Cover.

(12) United States Patent Svean et al.		(10) Patent No.: US 6,567,524 B1
		(45) Date of Patent: May 20, 2003
(54)	NOISE PROTECTION VERIFICATION DEVICE	4,548,082 A * 10/1985 Engebretson et al. 128/746 4,985,925 A 1/1991 Langberg et al. 5,201,006 A * 4/1993 Weinrich 381/312 5,276,739 A * 1/1994 Krokstad et al. 381/68.2 5,426,719 A 6/1995 Franks et al. 5,680,466 A * 10/1997 Zelikovitz 381/68.1
(75)	Inventors: Jarle Svean , Trondheim (NO); Svein Sørsdal , Trondheim (NO); Odd Kr. Ø. Pettersen , Trondheim (NO); Georg E. Ottesen , Trondheim (NO); Sverre Stensby , Trondheim (NO)	* cited by examiner <i>Primary Examiner</i> —Rexford Barnie
(73)	Assignee: Nacre AS , Trondheim (NO)	(74) <i>Attorney, Agent, or Firm</i> —Young & Thompson

Svean, Cover.

63. *Svean* recognizes that “it is critically important to avoid leakage of [noise] through or around the sealing and blocking parts of [a] hearing protector[.]”

Present day hearing protectors take the form either of sealing cups which enclose the ear, or ear plugs which blocks [sic] the ear canal. For both types, it is *critically important to avoid leakage of the noise sound through or around the sealing and blocking parts of the hearing protectors*.

Id., 3:39-43; *see also* 3:48-59, 3:60-67.

64. To address this recognized problem, *Svean* proposes a device capable of “in situ” verification that “a hearing protector is properly used.” *Id.*, 3:38-39.

It is also an object of this invention to provide a device for *verifying in situ that a hearing protector is properly used*.

Id., 3:38-39.

65. At 11:61-13:38, *Svean* discloses an embodiment of the invention directed to “Online verification/control of hearing protector performance.” This excerpt is reproduced below, which I will subsequently describe in more detail.

Another embodiment of the invention, called “*Online verification/control of hearing protector performance*[,]”[] utilises the fact that a sound field locally generated in the cavity near the eardrum is influenced by leakage in the hearing protector. A small electro-acoustic transducer (sound source) **SG** and a **microphone M2** are arranged in a **sealing section 2** arranged for attenuating sounds entering the meatus cavity 3. A **digital signal processing (DSP) or microcomputer unit E3** in the main section 1 or in the **sealing section 2** is used to generate a predetermined signal which is D/A converted by the D/A converter E7, amplified by amplifier E6 and applied to the **sound source SG**, which generates a sound field in the closed part of the meatus 3. The **microphone M2** picks up the sound in the meatus cavity 3. This signal is amplified by amplifier E4, A/D converted by A/D converter E5, and analysed in the **digital signal processor or microprocessor E3**. The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions. The user may get audible or other messaged confirmation if the leakage is acceptably low, or a warning signal if leakage is unacceptably high. In the same manner, a signal may be transmitted to other instances, e.g. an external industrial health monitoring unit, with information about the leakage. One example may be that an ear terminal according to the

invention is used for checking for leakage in the hearing protection while the wearer is at a gate controlling admittance to a noise exposed area. If leakage occurs, a signal may be transmitted from the ear terminal to a corresponding signal receiver at the gate, having means to block the gate for entrance until the leakage condition is remedied and verified.

FIG. 1 illustrates an embodiment of the invention where the verification device is integrated in a hearing protective earplug. This embodiment comprises an outer section 1 containing a **microphone M2** and a **sound generator SG**. An inner **sealing section 2** is attached; to the outer section, but may be made in one integrated outer section/ sealing section 1,2. An acoustic transmission channel T2 connects **sound generator SG** to the inner portion of the meatus 3. The **sound generator SG** produces a predetermined acoustic signal, which generates a sound field in the meatus 3. An acoustic transmission channel T1 connects **microphone M2** to the inner portion of the meatus 3. **Microphone M2** picks up the sound field being set up by the **sound generator SG**. The signal generation and analysis is carried out in a **digital signal processing (DSP) or microcomputer unit E3** with appropriate amplifiers and converters as described in the previous paragraphs. All the electronics 11 as well as the power supply 12 are provided in the outer section 1.

FIG. 8 illustrates a signal processing arrangement according to a preferred embodiment of the invention. This embodiment utilises a signal which produces reliable characterisation of the sound field in

the cavity, preferably while not being annoying to the user. The signal may comprise one or more sinusoidal components presented simultaneously, or in sequence. *Alternatively, a pseudorandom sequence may be employed.* In both cases, preferably both the in-phase and the out-of-phase portions of the sound field are analysed and used in the verification algorithm.

An example of the signal processing is shown in the flow diagram in FIG. 8. In the example, two pure tones of different frequencies f_1 and f_2 are generated by algorithms in the processing unit E3. The generators are represented by blocks 81 and 82 respectively. The generators generate both the in-phase (sin) and out-of-phase (cos) components. The in-phase components are added together in block 83, converted to analogue form, amplified and applied to the **sound generator SG**. The resulting sound field is picked up by the **microphone M2**, amplified, converted to digital form and analysed by algorithms in the **processing unit E3** for a series of detectors represented by blocks 84, 85, 86 and 87. The in-phase and out-of-phase components of the **microphone M2 signal** are analysed for each of the two frequencies. The detector algorithm performs a sample by sample multiplication of the two input signals and smoothes the result with a low-pass filter. *The four detector outputs are applied to a decision algorithm represented by block 88 where they are compared to stored values. The decision result may be a digital “go”/“no go” real time signal indicating acceptable noise protection attenuation or unacceptable protection conditions.* The result of the analysis is compared to stored results from previous

measurements of the same type in a situation with good sealing conditions.

The stored values for the decision algorithm may according to a preferred embodiment be based on previous laboratory experiments, but values for the decision algorithm may also be determined, e.g. making an average and setting a lower acceptance limit for a general-purpose embodiment of the invention.

The number and values of frequencies and the smoothing characteristics of the detectors are chosen as a compromise between audibility and response time. If a continuously running verification should be necessary, low frequencies, e.g. in the range of 10-20 Hz, of sufficiently low levels may be utilised in order to avoid annoyance. The pure tones may then be partially or fully aurally masked by the residual noise transmitted by the hearing protector.

The acoustic phenomenon on which the embodiment of the invention is based is illustrated by the electric analogy diagram in FIG. 9. In the diagram, the sound generator SG is modelled by its acoustic Thevenin equivalent represented by blocks 91 and 92. The sound pressure p_1 is generated by the Thevenin generator 91, resulting in a volume velocity through the Thevenin impedance $Z_1(f)$ 92. The microphone M2 is modelled by its acoustic impedance $Z_3(f)$ represented by block 93. The sound pressure p_2 at the microphone entrance is converted to an electric signal by the microphone. For the purpose of the present illustration, all [acoustic] elements exposed to the sound pressure, generated by the sound generator SG, except the microphone, are

lumped together in the acoustic impedance $Z_2(f)$ represented by block 95. A leakage in the hearing protector may be modelled by a change in the variable acoustic impedance $Z_2(f)$. The change will usually affect both the frequency dependent modulus and the frequency dependent phase of $Z_2(f)$. This change leads to a change in the relationship between the sound pressures p_2 and p_1 , which is analysed as described in connection with FIG. 8.

Svean, 11:61-13:38.

66. As disclosed in the excerpt above, *Svean* discloses that in the verification embodiment, a **processing unit E3** “is used to generate a predetermined signal[,] which is D/A converted ..., amplified...[,] and applied to the **sound source SG**, which generates a sound field in the closed part of the meatus 3.” *Id.*, 12:1-6. This sound is picked up by **microphone M2** in the meatus cavity 3. *Id.*, 12:6-7. This signal is amplified, A/D converted, and “analysed in the **digital signal processor or microprocessor E3**.” *Id.*, 12:7-9. “The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions,” and “[t]he user may get audible or other messaged confirmation if the leakage is acceptably low, or a warning signal if leakage is unacceptably high.” *Id.*, 12:9-14.

A **digital signal processing (DSP) or microcomputer unit E3** in the main section 1 or in the **sealing section 2** is used to generate a predetermined signal which is D/A converted by the D/A converter

E7, amplified by amplifier E6 and applied to the **sound source SG**, which generates a sound field in the closed part of the meatus 3. The **microphone M2** picks up the sound in the meatus cavity 3. This signal is amplified by amplifier E4, A/D converted by A/D converter E5, and analysed in the **digital signal processor or microprocessor E3**. The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions. The user may get audible or other messaged confirmation if the leakage is acceptably low, or a warning signal if leakage is unacceptably high.

Id., 12:1-14.

67. The following components shown in Fig. 1 of *Svean* are annotated below: **electronics unit 11**, **microphone M1**, **microphone M2**, **loud speaker/sound generator SG**, and **sealing section 2**.

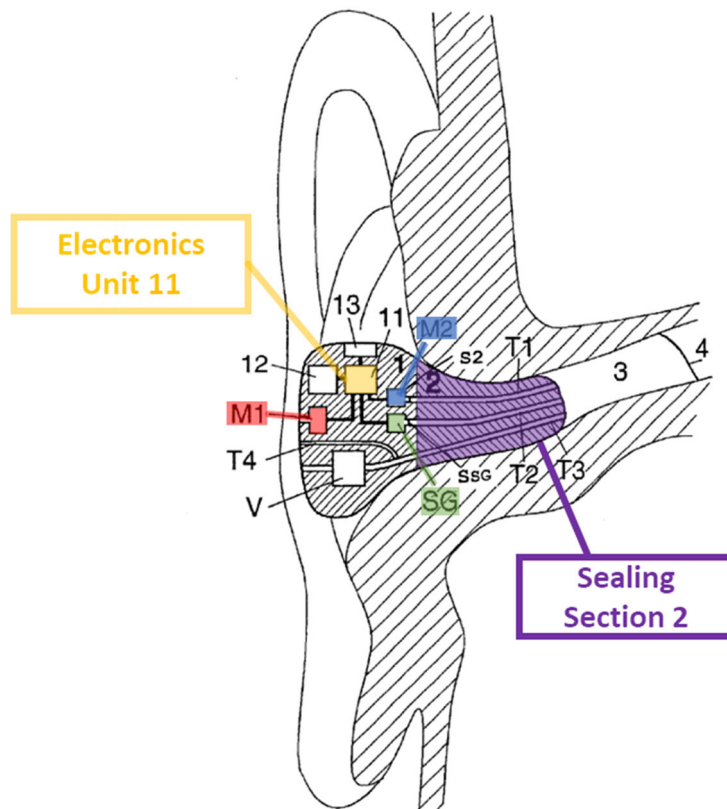


Fig. 1

Svean, Fig. 1.

68. *Svean* further discloses that the **processing unit E3** is connected to various **memory elements** that store program instructions and data, which is shown in Fig. 2.

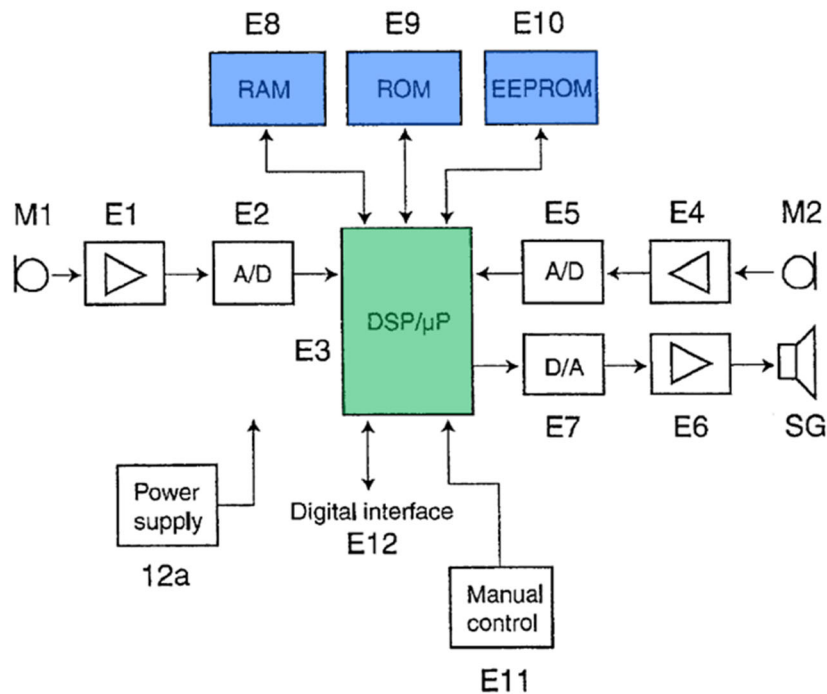


Fig. 2

The **processing unit E3** is connected to **memory elements RAM (Random access memory) E8, ROM (read only memory) E9, and EEPROM (electrically erasable programmable read only memory) E10**. The **memories E8, E9, and E10** are in a preferred embodiment of the invention used for *storing computer programs, filter coefficients, analysis data and other relevant data*.

Id., 7:23-29.

69. Figure 8 of *Svean* “illustrates a signal processing arrangement according to a preferred embodiment of the invention.” *Id.*, 12:42-43. *Svean* discloses that **processing unit E3** generates “a signal which produces reliable

characterisation of the sound field in the cavity, preferably while not being annoying to the user.” *Id.*, 12:43-45.

70. As one example, *Svean* discloses a test signal that comprises “one or more sinusoidal components”; alternatively, *Svean* discloses that the test signal may be a “pseudorandom sequence.” *Id.*, 12:42-50.

FIG. 8 illustrates a signal processing arrangement according to a preferred embodiment of the invention. This embodiment utilises a signal which produces reliable characterisation of the sound field in the cavity, preferably while not being annoying to the user. The signal may comprise one or more sinusoidal components presented simultaneously, or in sequence. ***Alternatively, a pseudorandom sequence may be employed.*** In both cases, preferably both the in-phase and the out-of-phase portions of the sound field are analysed and used in the verification algorithm.

Id., 12:42-50.

71. The signal processing illustrated in Figure 8 of *Svean* is for “two pure tones of different frequencies f_1 and f_2 .” *Id.*, 12:51-53. Figure 8 is shown below.

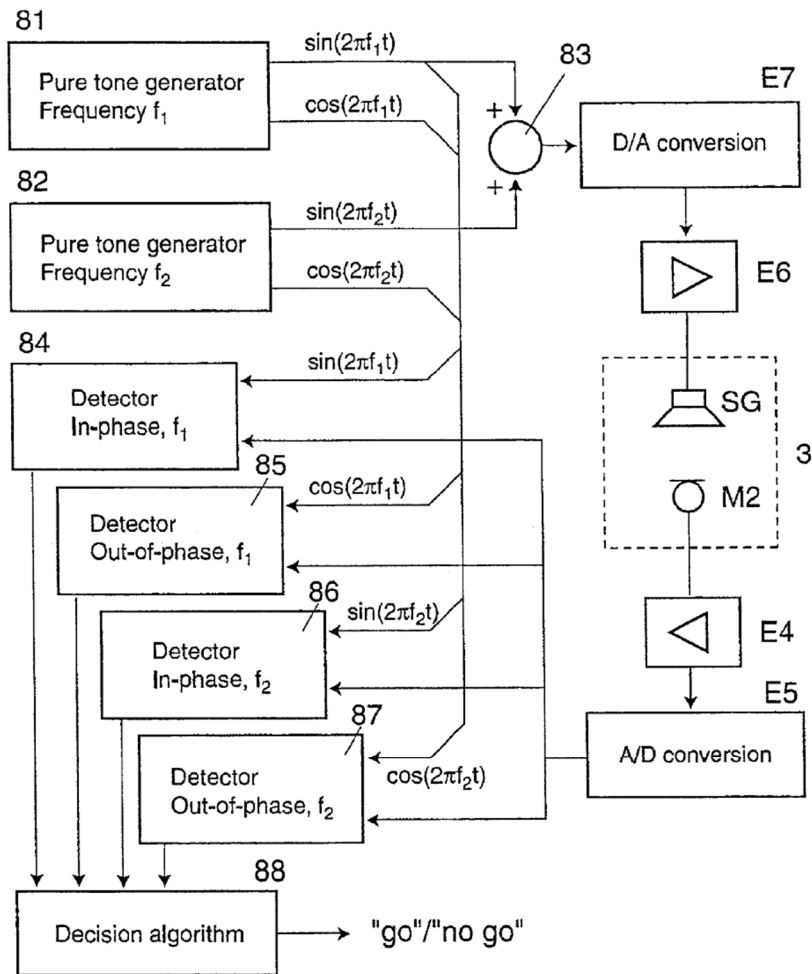


Fig. 8

Svean, Fig. 8.

72. *Svean* describes the analysis performed in this “two pure tone” embodiment at 12:51-13:6. *Svean* discloses generating the in-phase (sine) and out-of-phase (cos) components as to each frequency, adding the in-phase components and applying them to the sound generator, and analyzing the sounds picked up by **microphone M2**. *Id.*, __. The analysis performed in each of blocks 84 through 87

involves “a sample by sample multiplication of the two input signals” and “smooth[ing] the result with a low pass filter.” *Id.*, __. As to each block, the two input signals are: block 84 (the output of **microphone M2** and $\sin(2\pi f_1 t)$); block 85 (the output of **microphone M2** and $\cos(2\pi f_1 t)$); block 86 (the output of **microphone M2** and $\sin(2\pi f_2 t)$); and block 87 (the output of **microphone M2** and $\cos(2\pi f_2 t)$). *See id.*, Fig. 8.

73. The outputs of blocks 84 through 87 “are applied to a decision algorithm represented by block 88 where they are compared to stored values.” *Id.*, __; *id.* (“The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions.”).

74. *Svean* discloses that “[t]he decision result may be a digital ‘go’/‘no go’ real time signal indicating acceptable noise protection attenuation or unacceptable protection conditions.” *Id.*, __.

75. *Svean* discloses that “[t]he stored values for the decision algorithm may according to a preferred embodiment be based on previous laboratory experiments, but values for the decision algorithm may also be determined, e.g. making an average and setting a lower acceptance limit for a general-purpose embodiment of the invention.” *Id.*, 13:6-11.

76. As to the embodiment where the test signal comprises two pure tones, *Svean* discloses that “[t]he number and values of frequencies and the smoothing

characteristics of the detectors are chosen as a compromise between audibility and response time.” *Id.*, 13:12-14. “If a continuously running verification should be necessary, low frequencies, e.g. in the range of 10-20 Hz, of sufficiently low levels may be utilised in order to avoid annoyance.” *Id.*, 13:14-16. “The pure tones may then be partially or fully aurally masked by the residual noise transmitted by the hearing protector.” *Id.*, 13:16-18.

B. Voix

77. *Voix* is titled “method and apparatus for determining in situ the acoustic seal provided by an in-ear device.” *Voix*, Cover.

(12) United States Patent Voix et al.	(10) Patent No.: US 6,687,377 B2 (45) Date of Patent: Feb. 3, 2004
(54) METHOD AND APPARATUS FOR DETERMINING IN SITU THE ACOUSTIC SEAL PROVIDED BY AN IN-EAR DEVICE	FOREIGN PATENT DOCUMENTS WO WO 94/22372 * 10/1994 381/60 * cited by examiner
(75) Inventors: Jeremie Voix , Montreal (CA); Frederic Laville , Montreal (CA)	<i>Primary Examiner</i> —Xu Mei

Voix, Cover.

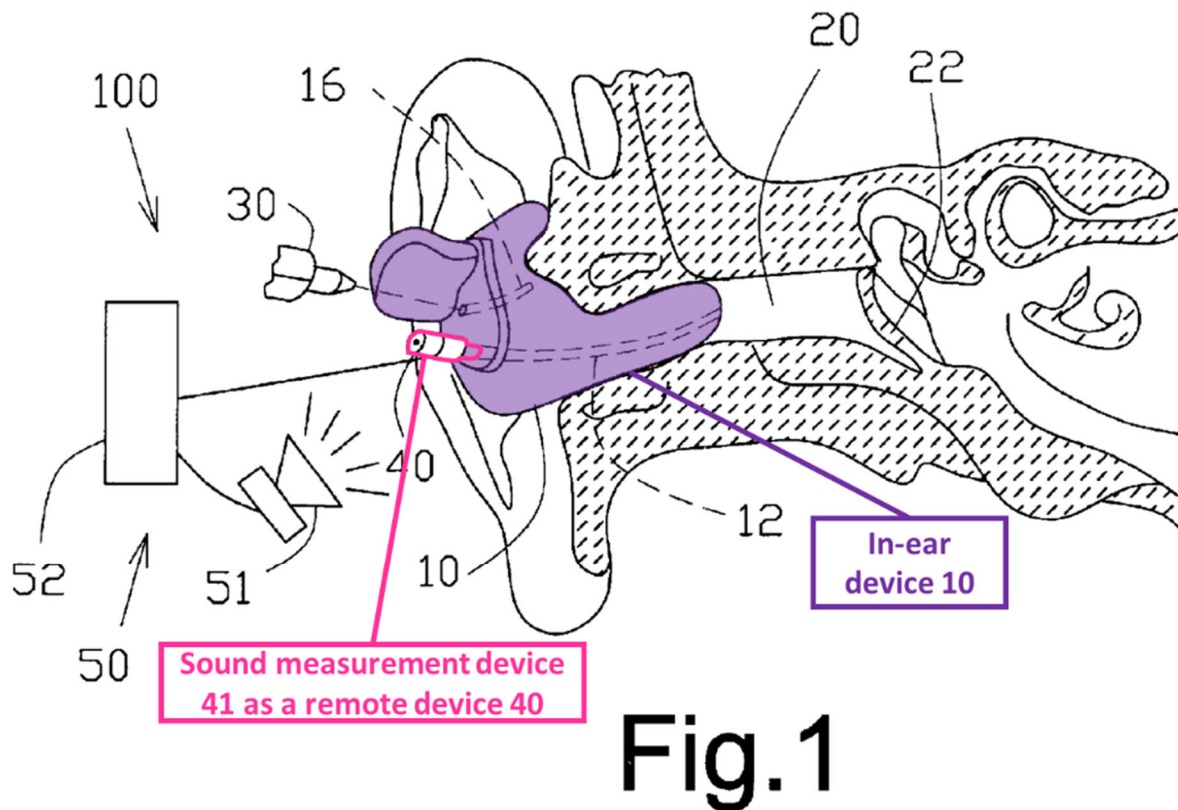
78. In the summary of the invention, *Voix* discloses that “[a]ccording to an aspect of the present invention, there is provided an apparatus for determining in situ an acoustic seal provided by an in-ear device inserted into an ear canal of an individual.” *Voix*, 3:33-36; *see also id.*, Abstract (“The present invention provides

an apparatus for determining in situ the acoustic seal provided by an in-ear device of a hearing protection/aid nature inserted into the ear canal of an individual.”).

According to an aspect of the present invention, there is provided an apparatus for determining in situ an acoustic seal provided by an in-ear device inserted into an ear canal of an individual, said in-ear device having a sound bore with an environment opening and an ear opening outside and inside said ear canal respectively, said environment opening being adapted to be removably engaged by a remote device such as a sound measurement device, a filter device, an amplifier device, a plug device and the like, said apparatus comprises a sound measurement device being said remote device and having a probe microphone and reference microphone isolated from each other and connected to a data processing unit having a control box and a reference sound source, both connected to a computer unit, said probe and reference microphones being adapted for measuring a sound pressure level inside said ear canal of said individual and a sound pressure level from said environment in close proximity of said in-ear device respectively, said sound pressure levels corresponding to a known noise signal created by said reference sound source at an inlet to the individual's ear in a close proximity of said in-ear device, said data processing unit being adapted for recording and processing said sound pressure levels read by both said probe and reference microphones corresponding to said known noise signal to provide a corresponding calculated value of a sound level difference provided by said in-ear device.

Voix, 3:33-58.

79. Fig. 1 discloses “an embodiment 100 of [the] apparatus for determining in situ the acoustic seal ...” *Id.*, 6:53-58. “The apparatus 100 comprises a **sound measurement device 41 as a remote device 40** ... that has a **probe microphone 42** and a **reference microphone 44** connected to a data processing unit 50 ...” *Id.*, 6:66-7:2. **Sound measurement device 41 as a remote device 40** is shown in more detail in Fig. 3.



Voix, Fig. 1.

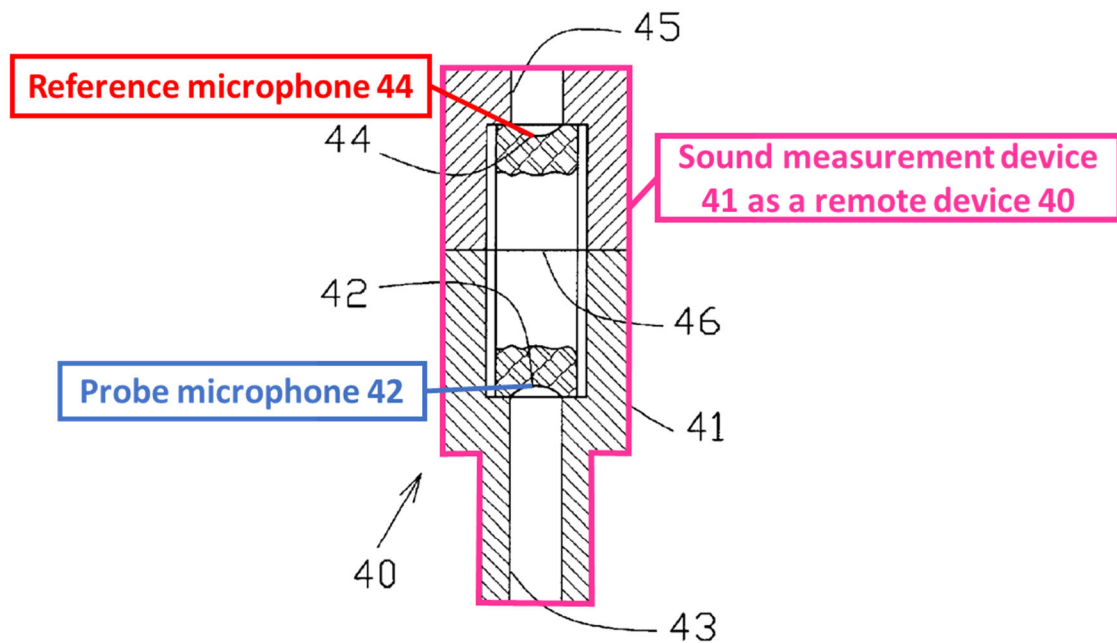


Fig.3

Voix, Fig. 3.

Referring to FIG. 1, there is shown *an embodiment 100 of an apparatus for determining in situ the acoustic seal*, preferably leading to an attenuation or an amplification of the sound level, provided by an **in-ear device 10**, preferably a semi-custom-molded expandable earplug, inserted into an ear canal 20 of an individual.

As it is shown in FIG. 2, the **in-ear device 10** has a sound bore 12 with an environment opening 13 and an ear opening 14 outside and inside the ear canal 20 respectively. The environment opening 13 is adapted to be removably engaged by a **remote device 40** such as a sound measurement device, a filter device, an hearing aid (or amplifier) device, a plug device and the like.

*The apparatus 100 comprises a **sound measurement device 41** as a **remote device 40** (see FIG. 3) that has a **probe microphone 42** and a **reference microphone 44** connected to a data processing unit 50 and isolated from each other by isolation barrier 46 located between the **probe 42** and **reference 44** microphones for fully canceling any mutual influence. Microphones **42**, **44** are typical microphones known in the industry such as EA-1954 or FG-3329 types from Knowles Electronics, Inc. Obviously, both microphones **42**, **44** are pre-calibrated with respect to their position relative to their respective opening 43, 45 of the **sound measurement device 41**. The barrier 46 is preferably made out of potting compound or epoxy like material being a sound insulating material securing each microphone **42**, **44** in place.*

Voix, 6:53-7:12.

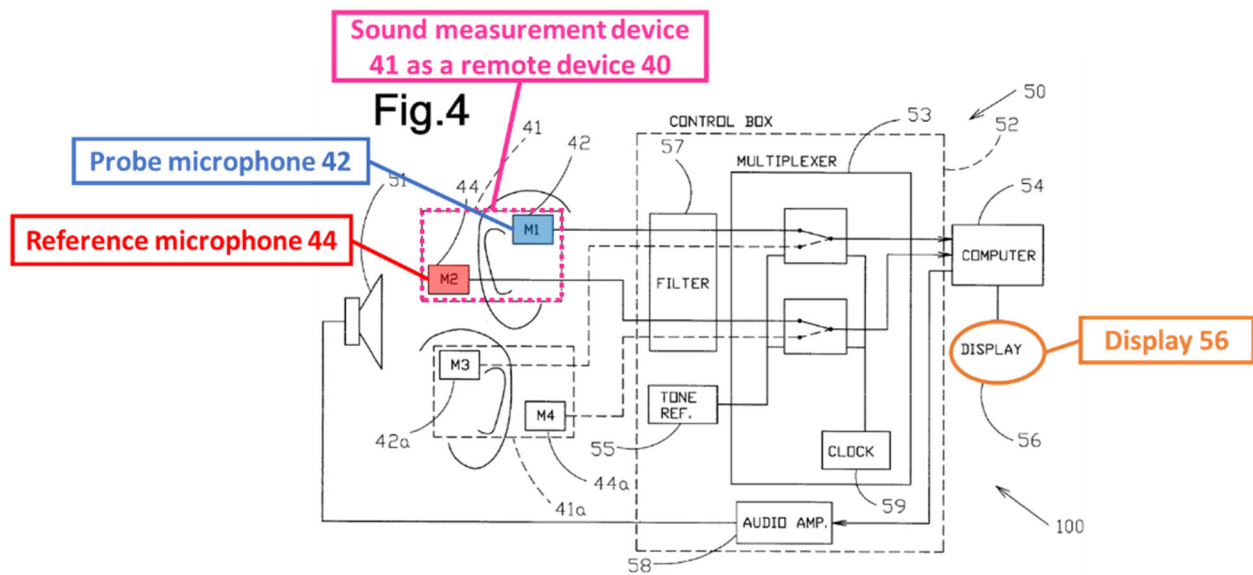
80. *Voix* discloses that the **probe microphone 42** and the **reference microphone 44** measure the sound pressure level inside the ear canal 20 and from the environment, respectively. *Id.*, 7:16-21. “The sound pressure levels correspond to a **known noise source signal, preferably of wide band ranging from twenty hertz (20 Hz) to twenty thousand hertz (20000 Hz) for enhanced measurements**, coming from the computer unit 54 via an audio amplifier 58 of the control box 52 and speaker 51.” *Id.*, 7:26-31. The control box 52 gathers the sound pressure levels and transmits them to the computer unit 54 to be recorded and processed. *Id.*, 7:31-35.

Referring to FIG. 4 the data processing unit 50 has a control box 52 and a reference sound source or speaker 51, both connected to a computer unit 54. *The probe 42 and the reference 44 microphones are adapted for measuring a sound pressure level inside the ear canal 20 of an individual and a sound pressure level from the environment in close proximity of the in-ear device 10 respectively* (see FIGS. 1 and 5). The sound pressure levels corresponding to a known noise pressure signal created by the reference sound source 51 at an inlet to the individual's ear in a close proximity of the in-ear device 10 and inside the individual's ear canal 20, in proximity to the tympanic membrane 22 next to the ear drum. *The sound pressure levels correspond to a known noise source signal, preferably of wide band ranging from twenty hertz (20 Hz) to twenty thousand hertz (20000 Hz) for enhanced measurements*, coming from the computer unit 54 via an audio amplifier 58 of the control box 52 and speaker 51. *The control box 52 is preferably an electronic controller adapted for gathering the sound pressure levels read by both microphones 42, 44 before transmitting them to the computer unit 54 to be recorded and processed by the same.* Although the use of a standard micro-computer or a "laptop" computer is preferred, the computer unit 54 could be any other electronic system such as a dedicated system or larger computers.

Voix, 7:14-38.

81. “[T]he sound pressure levels read by both **probe 42** and **reference 44** microphones corresponding to the known noise pressure signal” are processed⁴ to provide a corresponding *calculated value of a sound level difference* provided by the in-ear device 10.” *Id.*, 7:39-44. “The computer unit 54 preferably includes a **display member 56** for *displaying the sound level difference calculated value* provided by the **in-ear device 10**, being either an attenuation or an amplification depending on the type of **in-ear device 10**.” *Id.*, 7:44-48.

⁴ *Voix* refers to data processing unit 50 processing the sound pressure levels in this excerpt. *Id.*, 7:39-44. Whether computer unit 54 or data processing unit 50 performs the processing is not relevant because the grounds rely on *Svean*’s processing.



The data processing unit 50 is adapted for recording and processing the sound pressure levels read by both **probe 42** and **reference 44** microphones corresponding to the known noise pressure signal to provide a corresponding *calculated value of a sound level difference provided by the in-ear device 10*. The computer unit 54 preferably includes a **display member 56** for *displaying the sound level difference calculated value* provided by the **in-ear device 10**, being either an attenuation or an amplification depending on the type of **in-ear device 10**.

Voix, 7:38-48.

82. Consistent with the description above, *Voix* discloses a method utilizing these components to determine the acoustic seal. *Id.*, 4:50-6:22; 8:63-10:20.

C. *Padhi*

83. *Padhi* is titled “apparatus, method, and computer program for comparing audio signals.” *Padhi*, Cover.

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(12) Patent Application Publication	(43) Pub. Date: May 5, 2005
Padhi et al.	
(54) APPARATUS, METHOD, AND COMPUTER PROGRAM FOR COMPARING AUDIO SIGNALS	(22) Filed: Nov. 4, 2003
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84. *Padhi* discloses “[a]n apparatus, method, and computer program [that] are capable of receiving and cross-correlating a first audio signal and a second audio signal.” *Padhi*, Abstract.

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

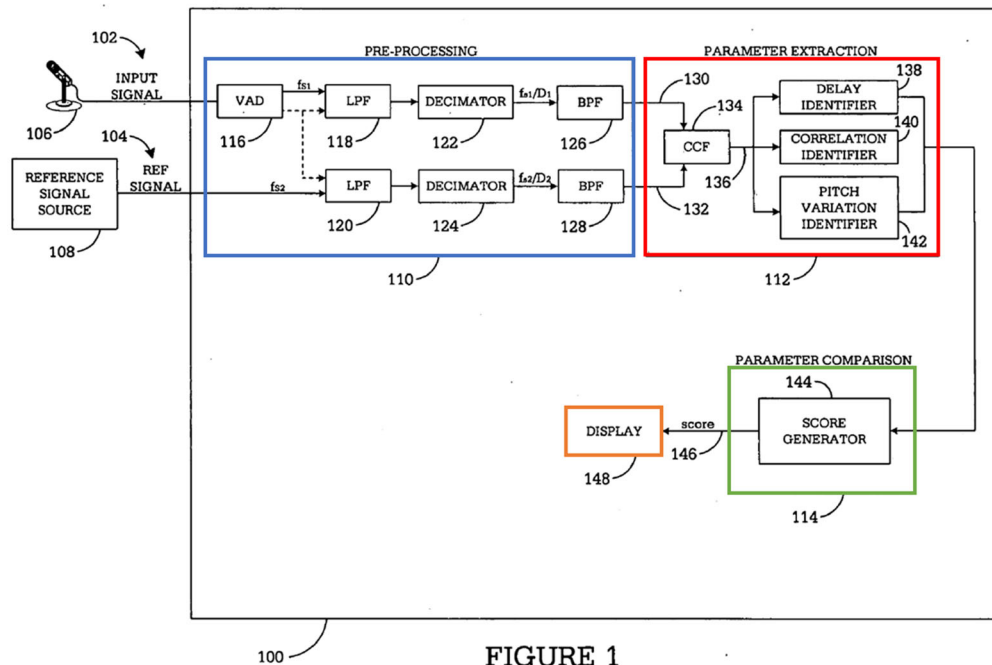


FIGURE 1

Padhi, Figure 1.

As shown in FIG. 1, the apparatus 100 includes a **pre-processing section 110**, a **parameter extraction section 112**, and a **parameter comparison section 114**. In general, the **pre-processing section 110** prepares the signals **102**, **104** for comparison, the **parameter extraction section 112** identifies characteristics of the signals **102**, **104**, and the **parameter comparison section 114** uses the characteristics to generate a score.

Padhi, [0020].

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].

86. *Padhi* discloses that the various components in Figure 1 “may represent software routines executed by one or more processors.” *Id.*, [0034]. Further, *Padhi* discloses 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.” *Id.*, [0035].

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].

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].

87. *In Padhi*, the apparatus of Figure 1 receives an **input audio signal 102** via a microphone. *Id.*, [0018]. A **reference signal 104** is supplied by a reference signal source. *Id.* The signals “could be replaced by any suitable audio signals provided by any suitable source or sources.” *Id.*

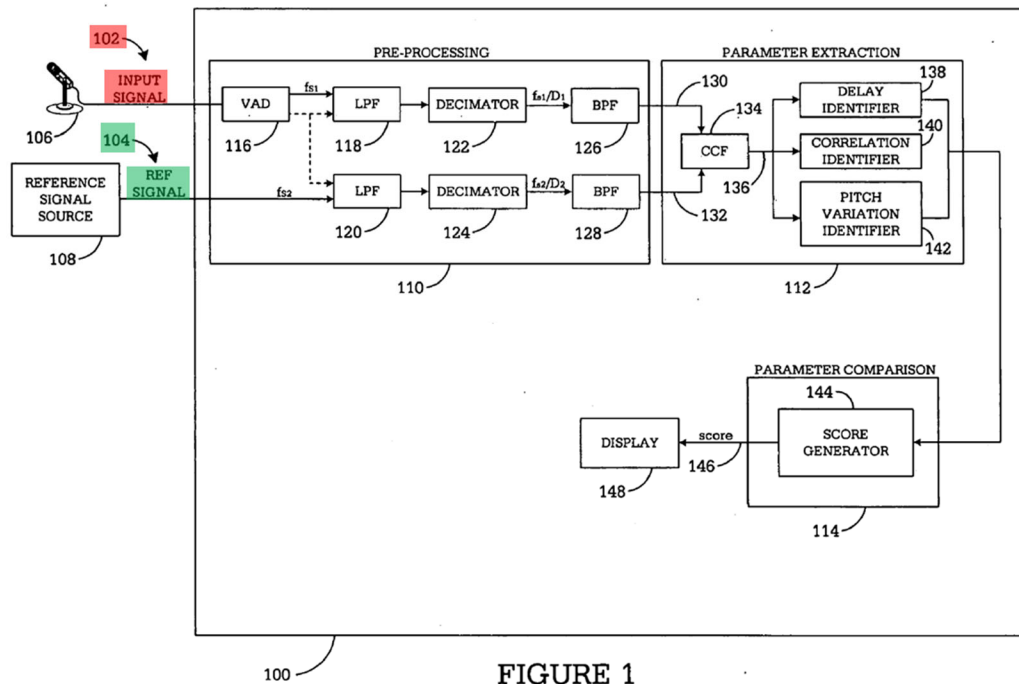


FIGURE 1

Padhi, Fig. 1.

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].

88. 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. *Id.*, [0023], [0025], [0071].

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].

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].

89. The apparatus cross-correlates the two signals to determine the extent to which they match. *Id.*, [0017], [0027], [0028], [0030].

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 **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].

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].

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].

90. The amount of correlation between the signals is a component of an overall “score,” which may also include the amount of delay and variation in pitch between the signals. *Id.*, [0032]. The apparatus then presents the score to an end user via a display. *Id.*, [0033].

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 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].

91. *Padhi* discloses more details regarding an example technique for generating a score with reference to Figure 6. *Id.*, [0069]-[0078].

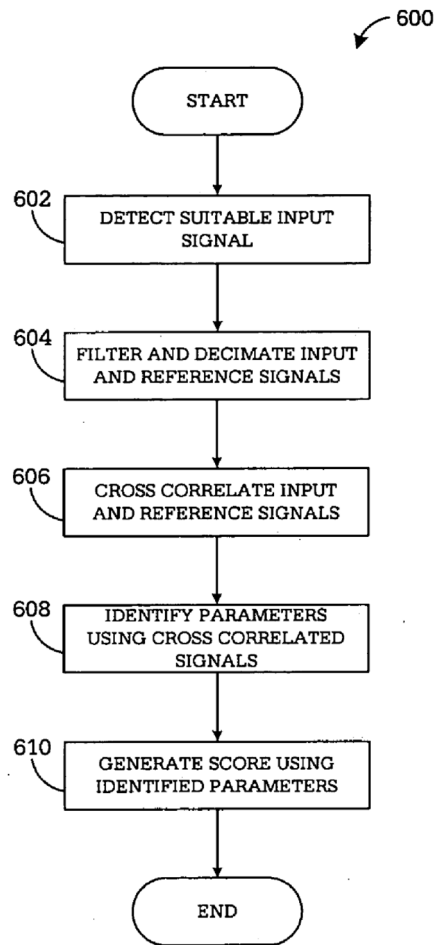


FIGURE 6

Padhi, Fig. 6.

FIG. 6 illustrates an example method 600 for comparing audio signals according to one embodiment of this disclosure. For ease of explanation, the method 600 is described with respect to the audio apparatus 100 of FIG. 1. The method 600 could be used by any other suitable apparatus or system.

Padhi, [0069].

92. At step 602, “apparatus 100 detects a suitable input signal.” *Id.*, [0070]. At step 604, “apparatus 100 filters and decimates the input and reference signals.” *Id.*, [0071]. “This may include, for example, the filters 118, 120 performing low pass filtering” on the input signal 102 and the reference signal 104. *Id.* At step 606, “apparatus 100 cross-correlates the processed signals.” *Id.*, [0072]. “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.” *Id.*

The apparatus 100 detects a suitable input signal at step 602. This may include, for example, the voice activity detector 116 receiving an **input signal 102** and determining that the **input signal 102** contains audio information representing a voice. This may also include the voice activity detector 116 instructing the filters 118, 120 to begin processing the **input signal 102** and a **reference signal 104**.

Padhi, [0070].

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].

93. At step 608, “apparatus 100 identifies a plurality of parameters using the cross-correlated signals.” *Id.*, [0073]. “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.” *Id.*

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].

94. At step 610, “apparatus 100 uses the identified parameters to generate a score.” *Id.*, [0074]. “The score identifies the extent to which the signals 102, 104 match.” *Id.* For calculating the score, score generator 144 “may use any suitable technique.” *Id.*, [0075]. For example, “[i]n some embodiments, the score generator 144 assigns different weights or levels of importance to the values produced by the identifiers 138-142.” *Id.*

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].

The score generator 144 may use any suitable technique to generate a score using the parameters. In some embodiments, the score generator 144 assigns different weights or levels of importance to the values produced by the identifiers 138-142. For example, when used in a karaoke system, the operator of the apparatus 100 may want a singer who incorrectly sings the lyrics of a song to receive a lower score 146 than a singer who sings at a variable pitch. In this example, a higher weight is assigned to singing the correct words, which can be determined using the delay and correlation values produced by the identifiers 138-140. The weights can be assigned in any suitable

manner, such as having default values and allowing a user to assign custom values as desired.

Padhi, [0075].

95. *Padhi* describes particular embodiments where the score generator “uses a two-stage scoring algorithm to determine a score 146.” *Id.*, [0076]. For example, the score generator 144 “may generate a first score using the delay value and correlation value produced by the delay identifier 138 and correlation identifier 140.” *Id.* The score generator 144 “may then generate a second score using the pitch variation value produced by the pitch variation identifier 142.” *Id.* “The final score 146 may represent a sum of the first and second scores.” *Id.*

In particular embodiments, the score generator 144 uses a two-stage scoring algorithm to determine a score 146. For example, the score generator 144 may generate a first score using the delay value and correlation value produced by the delay identifier 138 and correlation identifier 140. The score generator 144 may then generate a second score using the pitch variation value produced by the pitch variation identifier 142. The final score 146 may represent a sum of the first and second scores.

Padhi, [0076].

96. In paragraphs [0077]-[0082], *Padhi* describes more details regarding the example two-stage algorithm. “First, the score generator 144 generates a value

α using information from the delay identifier 138 and the correlation identifier 140.” *Id.*, [0077]. Score generator 144 generates the value α “using information from the delay identifier 138 and the correlation identifier 140.” *Id.* The value is determined according to the formula $\text{local_maxout}/\text{max_out}$. *Id.* In this calculation, “local_maxout identifies the largest value of the cross-correlated signal within a certain range...” *Id.*, [0078]. “[M]ax_out identifies the largest value in the cross-correlated signal.” *Id.* The correlation identifier 140 outputs the two values for local_maxout and max_out, and “[u]sing the value of α , the score generator 144 selects a first score (Score_1) using Table 1.” *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 α using the information from the delay identifier 138 and the correlation identifier 140. The value of α 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 (± 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 α , the score generator 144 selects a first score (Score₁) using Table 1.

Padhi, [0078].

97. Table 1 is shown below.

TABLE 1

Range	Score ₁
$\alpha \geq 0.75$	60
$0.75 > \alpha \geq 0.5$	50
$0.5 > \alpha \geq 0.25$	40
$0.25 > \alpha$	30

Padhi, [0078], Table 1.

98. As I previously noted, *Padhi* discloses that “[t]he score generator 144 may use any suitable technique to generate a score using the parameters.” *Id.*, [0075]. Thus, a person having ordinary skill in the art would have understood that *Padhi* discloses that Score₁ alone may be used as the final score.

99. However, in the particular embodiment disclosed in *Padhi*, *Padhi* discloses that “the score generator 144 uses the information from the pitch variation identifier 142 to select a second score, which is added to the first score to produce the final score 146.” *Id.*, [0079]. “In particular, the score generator 144 produces two values β_1 and β_2 ,” and *Padhi* provides formulas for these calculations in equations (7) and (8). *Id.* “Using the values of β_1 and β_2 , the score generator 144

assigns a final score using Table 2.” *Id.*, [0080]. “The formula adds the first score to a second score and produces the final score 146.” *Id.*

After that, the score generator 144 uses the information from the pitch variation identifier 142 to select a second score, which is added to the first score to produce the final score 146. In particular, the score generator 144 produces two values β_1 and β_2 , which uses the estimated coincidental harmonic frequency for three successive frames 202 calculated using Equation 5. The score generator 144 uses the following equations to identify values for β_1 and β_2 :

$$\beta_1 = \left| \frac{Pitch_{est}^{l-1} - Pitch_{est}^{l-2}}{Pitch_{est}^{l-2}} \right| \quad (7)$$

$$\beta_2 = \left| \frac{Pitch_{est}^l - Pitch_{est}^{l-1}}{Pitch_{est}^{l-1}} \right| \quad (8)$$

Padhi, [0079].

Ideally, if the **input signal 102** has a constant pitch, the deviation of the estimated coincidental harmonic frequency across the frames 202 should be small. Using the values of β_1 and β_2 , the score generator 144 assigns a final score using Table 2. In Table 2, if both the “Range 1” condition and “Range 2” condition in the same line are satisfied, the final score is computed using the formula on that line. The formula adds the first score to a second score and produces the final score 146. In Table 2, the symbol denotes the logical OR operation.

Padhi, [0080].

100. Table 2 is shown below.

TABLE 2		
Range 1	Range 2	Final Score
$\beta_1 < 0.2$	$\beta_2 < 0.2$	$\text{Score}_1 + 40$
$\beta_1 \parallel \beta_2 < 0.2$	$\beta_1 \parallel \beta_2 < 0.3$	$\text{Score}_1 + 20$
$\beta_1 \parallel \beta_2 < 0.3$	$\beta_1 \parallel \beta_2 < 0.6$	Score_1
$\beta_1 \parallel \beta_2 \geq 0.3$	$\beta_1 \parallel \beta_2 < 0.6$	$\text{Score}_1 - 10$
$\beta_1 \geq 0.6$	$\beta_2 \geq 0.6$	$\text{Score}_1 - 30$

Padhi, [0080], Table 2.

101. As previously discussed, *Padhi* discloses that the final score 146 “is presented to a user or other person or people on a display 148.” *Id.*, [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.

Id.

D. Hayashi

102. *Hayashi* is titled “Leak tester of ear-mounted external sound processing device.” *Hayashi*, Cover.

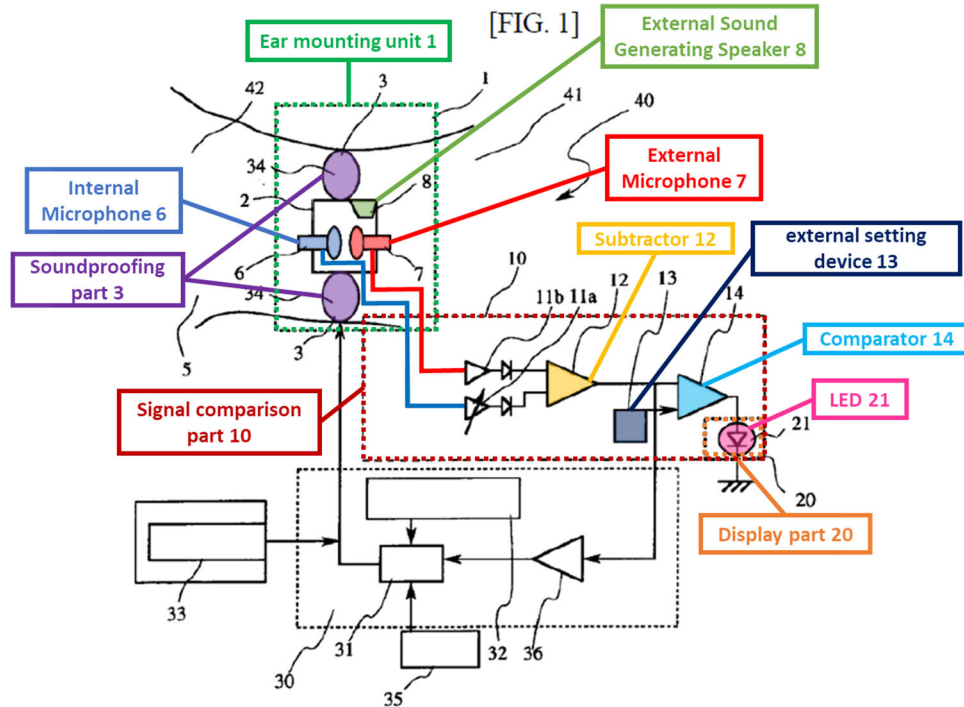
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		Examination Request: Not Yet		Total No. of Claims: 7	
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		Continued on final page			
(54) [Title of Invention] LEAK TESTER OF EAR-MOUNTED EXTERNAL SOUND PROCESSING DEVICE					

Hayashi, Cover.

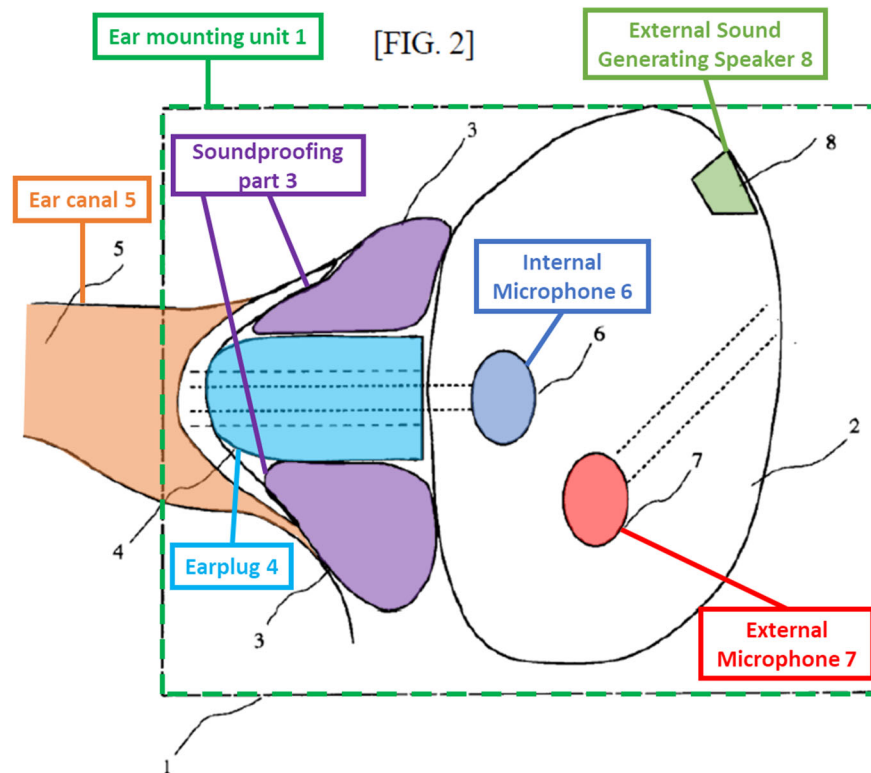
103. *Hayashi* discloses an **ear mounting unit 1** and a technique to test its seal. The technique involves subtracting microphone signals received by an **internal microphone 6** and an **external microphone 7** that receive sounds from opposite sides of a **soundproofing part 3**.

104. Specifically, the technique involves subtracting the two signals (**subtractor 12**) and comparing the result (**comparator 14**) to a threshold set by **external setting device 13**. *Hayashi* further discloses displaying the result of the seal test using **display part 20/LED 21**. *Id.*, [0010]-[0014], [0018]-[0020].

105. The components discussed above, as well as certain additional components/elements of *Hayashi's* Figures 1 and 2, are annotated below.



Hayashi, Fig. 1.



Hayashi, Fig. 2.

[Embodiments of the Invention] A leak tester of an embodiment of the present invention will be described with reference to FIGS. 1 and 2. FIG. 1 is a block diagram, and FIG. 2 is an enlarged diagram of when a mounting unit is inserted into an **ear canal**. The leak tester is composed of an **ear mounting unit 1**, a **signal comparison part 10**, a **display part 20**, and soundproofing part deformation means 30. Furthermore, the **ear mounting unit 1** is composed of a housing part 2, a **soundproofing part 3**, and an **earplug 4**.

Hayashi, [0010].

The housing part 2 of the **ear mounting unit 1** has an **external sound generating speaker 8**, an **internal microphone 6**, and an **external**

microphone 7, and the **soundproofing part 3** is ring-shaped and is fitted and closely attached around the earplug 4. In FIG. 1, from the housing part 2, the left side is an interior of an **ear canal 5**, that is, a soundproofing space 42, and the right side is an external space, that is, a free sound field 41. External sound 40 generated in the free sound field 41 is collected by the **external microphone 7**. Furthermore, the external sound 40 is soundproofed by the **soundproofing part 3** and the **earplug 4**, but components of the external sound 40 which are not completely soundproofed are collected in the **internal microphone 6**. The external sound 40 collected from each microphone is converted into an electrical signal and inputted to the **signal comparison part 10**.

Hayashi, [0011].

The **soundproofing part 3** is configured by, for example, a rubber or resin ring-shaped tube, and is provided with a space part in the interior, and the internal space is connected to the soundproofing part deformation means 30. The soundproofing part deformation means 30 supplies a soundproofing material 34 of silicone resin to the internal space of the soundproofing part 3. Here, the soundproofing material 34 is composed of, for example, a two-component curing silicone rubber or a UV-curing elastic resin, and is a composition in which a soundproofing agent 32 and a curing agent 33 are mixed. The mixed soundproofing material 34 is injected into the internal space of the soundproofing part 3 by the soundproofing part deformation means 30 so as to fit the shape of the ear canal 5. Because the soundproofing

material 34 (soundproofing agent 32 and curing agent 33) is made to start curing after a predetermined time (for example, about 5 minutes) has elapsed after mixing, the soundproofing agent 34 is injected from the soundproofing part deformation means 30 within the predetermined time until the sound pressure difference between the **ear canal 5** and the free sound field 41 reaches a predetermined value, and after achieving the sound pressure difference, injection of the soundproofing material 34 is stopped and the shape is fixed after the predetermined time elapses.

Hayashi, [0012].

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].

Furthermore, with the present invention, it is possible to acquire a shape of the **soundproofing part 3** such that a target soundproofing state is obtained. First, the threshold of the **external setting device 13** is set to correspond to a predetermined sound pressure difference, and the presentation sound is changed and generated. The presentation sound is converted into an electrical signal by the **internal microphone 6** and the **external microphone 7** to become a chemical injection pump drive signal of the soundproofing part deformation means 30 via the **signal comparison part 10**, and the soundproofing material 34 is injected into the **soundproofing part 3**. This operation is continued until the sound pressure difference reaches a predetermined value, and when the predetermined value is achieved, injection of the soundproofing material 34 into the **soundproofing part 3** is stopped. Note that the soundproofing material 34 may be automatically injected until the sound pressure difference reaches the threshold of the **external setting device 13** by adjusting the gain of the amplifier 36 using the setting value of the **external setting device 13** of the **signal comparison part 10**. Thus, the shape of the **soundproofing part 3** may be acquired.

Hayashi, [0019].

[Effect of the Invention] According to this invention, a device having a function of objectively measuring and displaying a leakage amount of an external sound can be provided. Furthermore, an external sound processing device such as a hearing aid having a function for displaying the leakage amount of sound, and having a small size suitable for a shape of an ear of a user and having a high soundproofing capability can be acquired.

Hayashi, [0020].

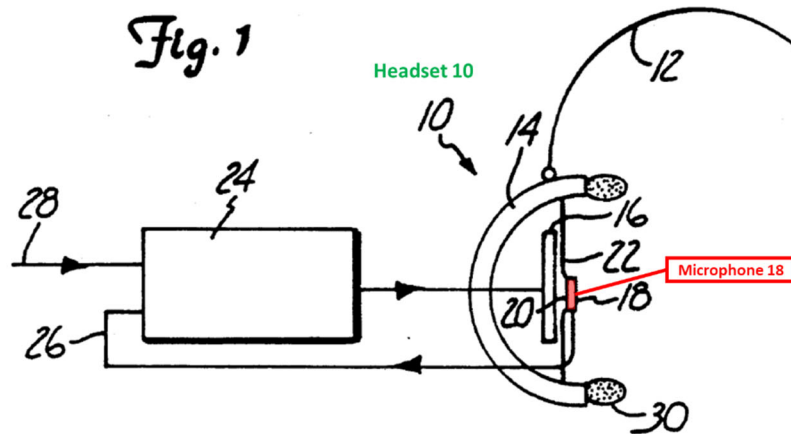
E. *Bourk*

106. *Bourk* is titled “noise cancellation headset.” *Bourk*, Cover.

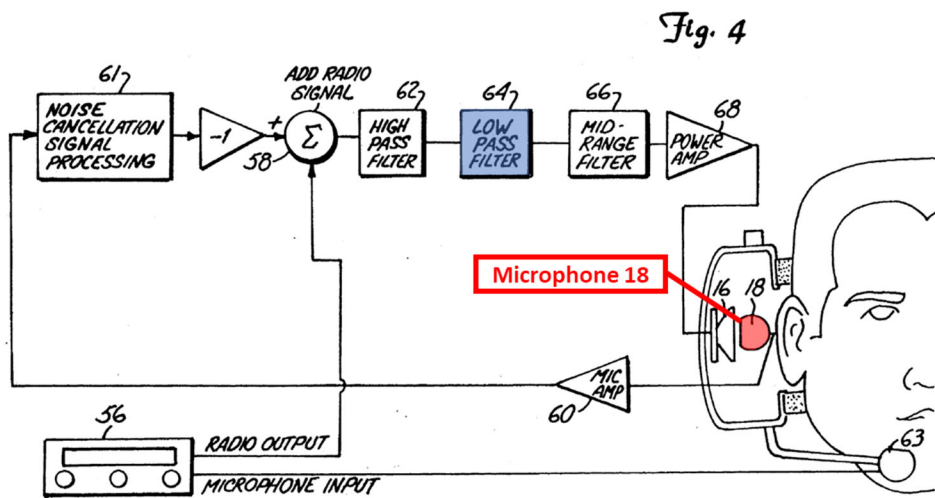
	
US005182774A	
United States Patent [19]	[11] Patent Number: 5,182,774
Bourk	[45] Date of Patent: Jan. 26, 1993
[54] NOISE CANCELLATION HEADSET	tion,” National Technical Information Service, U.S. Department of Commerce, Apr., 1975, pp. 86-87 117-119.
[75] Inventor: Terrance R. Bourk , San Diego, Calif.	<i>Primary Examiner</i> —Curtis Kuntz
[73] Assignee: Telex Communications, Inc. , Minneapolis, Minn.	<i>Assistant Examiner</i> —William Cumming
[21] Appl. No.: 555,990	<i>Attorney, Agent, or Firm</i> —Fredrikson & Byron

Bourk, Cover.

107. 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.

108. 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.

109. *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.

110. *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.

IX. CLAIM CONSTRUCTION

111. 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.”

EX1021, 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.”

Id., 39, 50.

112. 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 35 U.S.C. §112(f) as “means-plus-function” limitation. EX1017, 7-8. Patent Owner has appealed and disputed that the term is subject to §112(f). EX1023, 13-24.

113. 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.

114. However, with respect to the term “processor” as it appears in the ’493 patent’s claims, I have been asked to identify in the specification of the ’493 patent a processor or equivalent component that performs an algorithm corresponding to the operations attributed to the claimed “processor.”

115. Accordingly, I note that the ’493 patent discloses a microprocessor 206 or processor 206. More particularly, “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.” ’493 patent, 5:63-67.

116. Furthermore, I note that processor 206 performs an algorithm that is depicted in Figures 3 and 4 of the patent. In particular, as I explain further below, the algorithm includes the following steps: 304 (generating a test signal), 306 (emitting the test signal from a speaker), 308 (obtaining an ear canal microphone signal), 310/412 (calculating a value using a cross-correlation), 414 (comparing the calculated value to a threshold), 418/420 (generating and sending a message to a user). Figure 14, moreover, depicts an algorithm that is applicable to cross-correlation, correlation, and coherence.

117. Further, 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).

X. SPECIFIC GROUNDS FOR CHALLENGE

A. Ground 1: *Svean*, *Voix*, and *Padhi* Render Obvious Claims 1, 2, 4-8, 10, 11, 13, 14, 18-20

118. I have reviewed *Svean* and *Padhi*, and as described below, it is my opinion that *Svean* in combination with *Padhi* renders obvious all of the features of claims 1, 2, 4-8, 10, 11, 13, 14, 18-20.

1. Claim 1

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

119. 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 *Svean* discloses or suggests this feature.

120. *Svean*’s “ear terminal” shown, for example, in Fig. 1, corresponds to the claimed “audio device configured to test a seal of an earphone.” *Svean*’s ear terminal includes **memory** and a **processing unit E3** that is configured to execute instructions to perform operations that test the seal of the ear terminal. *See* §§X.A.1.a-X.A.1.c(vi).

121. At 11:61-13:38, *Svean* discloses methods and systems for “[o]nline verification/control of hearing protector performance,” which “utilises [sic] the fact

that a sound field locally generated in the cavity near the eardrum is influenced by leakage in the hearing protector.” *Id.*, 11:61-65.

Another embodiment of the invention, called “**Online verification/control of hearing protector performance**[,]”[] utilises the fact that a sound field locally generated in the cavity near the eardrum is influenced by leakage in the hearing protector. A small electro-acoustic transducer (sound source) **SG** and a **microphone M2** are arranged in a **sealing section 2** arranged for attenuating sounds entering the meatus cavity 3. A **digital signal processing (DSP) or microcomputer unit E3** in the main section 1 or in the **sealing section 2** is used to generate a predetermined signal which is D/A converted by the D/A converter E7, amplified by amplifier E6 and applied to the **sound source SG**, which generates a sound field in the closed part of the meatus 3. The **microphone M2** picks up the sound in the meatus cavity 3. This signal is amplified by amplifier E4, A/D converted by A/D converter E5, and analysed in the **digital signal processor or microprocessor E3**. The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions. The user may get audible or other messaged confirmation if the leakage is acceptably low, or a warning signal if leakage is unacceptably high. In the same manner, a signal may be transmitted to other instances, e.g. an external industrial health monitoring unit, with information about the leakage. One example may be that an ear terminal according to the invention is used for checking for leakage in the hearing protection

while the wearer is at a gate controlling admittance to a noise exposed area. If leakage occurs, a signal may be transmitted from the ear terminal to a corresponding signal receiver at the gate, having means to block the gate for entrance until the leakage condition is remedied and verified.

FIG. 1 illustrates an embodiment of the invention where the verification device is integrated in a hearing protective earplug. This embodiment comprises an outer section 1 containing a **microphone M2** and a **sound generator SG**. An inner **sealing section 2** is attached; to the outer section, but may be made in one integrated outer section/ sealing section 1,2. An acoustic transmission channel T2 connects **sound generator SG** to the inner portion of the meatus 3. The **sound generator SG** produces a predetermined acoustic signal, which generates a sound field in the meatus 3. An acoustic transmission channel T1 connects **microphone M2** to the inner portion of the meatus 3. **Microphone M2** picks up the sound field being set up by the **sound generator SG**. The signal generation and analysis is carried out in a **digital signal processing (DSP) or microcomputer unit E3** with appropriate amplifiers and converters as described in the previous paragraphs. All the electronics 11 as well as the power supply 12 are provided in the outer section 1.

FIG. 8 illustrates a signal processing arrangement according to a preferred embodiment of the invention. This embodiment utilises a signal which produces reliable characterisation of the sound field in the cavity, preferably while not being annoying to the user. The signal

may comprise one or more sinusoidal components presented simultaneously, or in sequence. *Alternatively, a pseudorandom sequence may be employed.* In both cases, preferably both the in-phase and the out-of-phase portions of the sound field are analysed and used in the verification algorithm.

An example of the signal processing is shown in the flow diagram in FIG. 8. In the example, two pure tones of different frequencies f_1 and f_2 are generated by algorithms in the processing unit E3. The generators are represented by blocks 81 and 82 respectively. The generators generate both the in-phase (sin) and out-of-phase (cos) components. The in-phase components are added together in block 83, converted to analogue form, amplified and applied to the **sound generator SG**. The resulting sound field is picked up by the **microphone M2**, amplified, converted to digital form and analysed by algorithms in the **processing unit E3** for a series of detectors represented by blocks 84, 85, 86 and 87. The in-phase and out-of-phase components of the **microphone M2 signal** are analysed for each of the two frequencies. The detector algorithm performs a sample by sample multiplication of the two input signals and smoothes the result with a low-pass filter. *The four detector outputs are applied to a decision algorithm represented by block 88 where they are compared to stored values. The decision result may be a digital “go”/“no go” real time signal indicating acceptable noise protection attenuation or unacceptable protection conditions.* The result of the analysis is compared to stored results from previous

measurements of the same type in a situation with good sealing conditions.

The stored values for the decision algorithm may according to a preferred embodiment be based on previous laboratory experiments, but values for the decision algorithm may also be determined, e.g. making an average and setting a lower acceptance limit for a general-purpose embodiment of the invention.

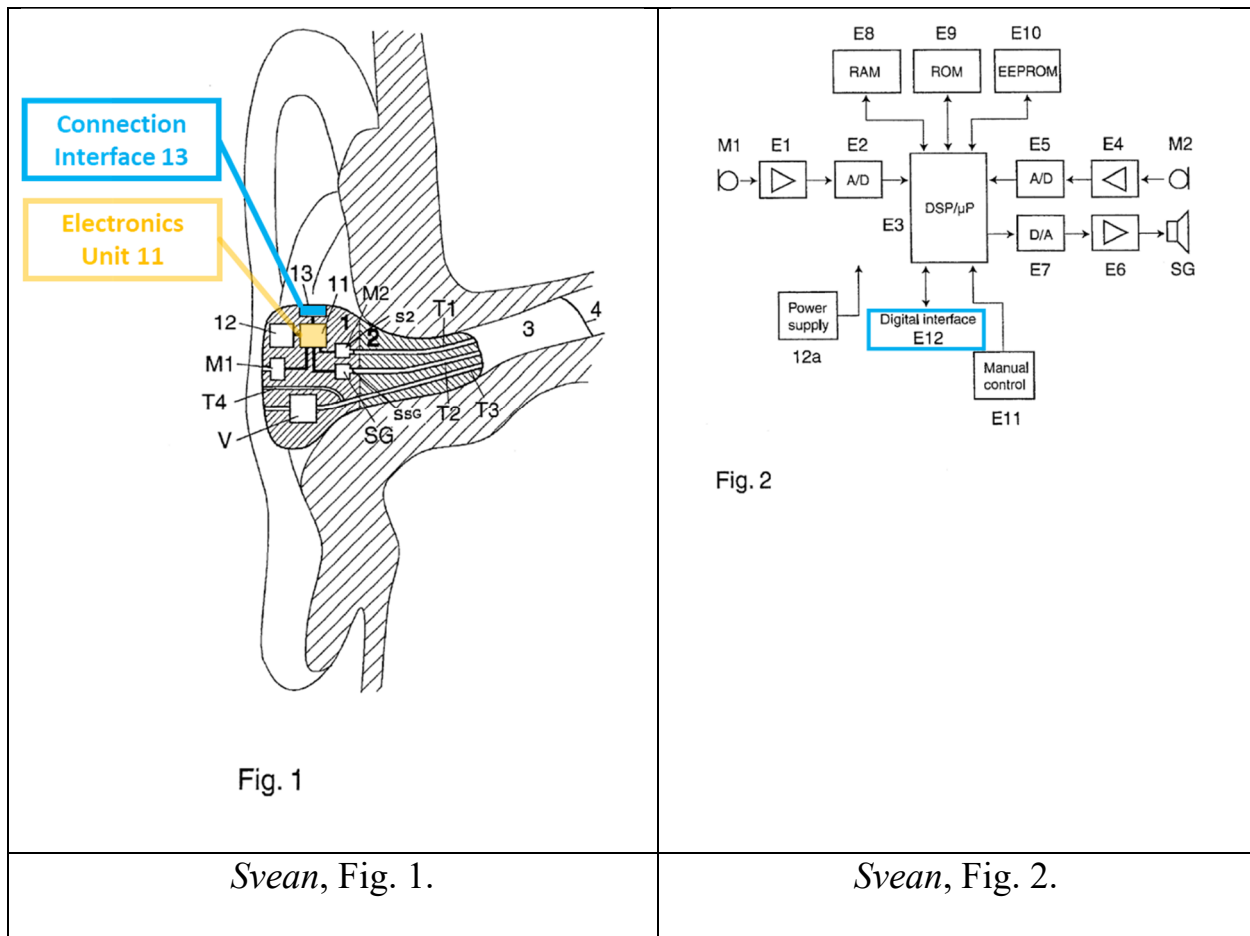
The number and values of frequencies and the smoothing characteristics of the detectors are chosen as a compromise between audibility and response time. If a continuously running verification should be necessary, low frequencies, e.g. in the range of 10-20 Hz, of sufficiently low levels may be utilised in order to avoid annoyance. The pure tones may then be partially or fully aurally masked by the residual noise transmitted by the hearing protector.

The acoustic phenomenon on which the embodiment of the invention is based is illustrated by the electric analogy diagram in FIG. 9. In the diagram, the sound generator SG is modelled by its acoustic Thevenin equivalent represented by blocks 91 and 92. The sound pressure p_1 is generated by the Thevenin generator 91, resulting in a volume velocity through the Thevenin impedance $Z_1(f)$ 92. The microphone M2 is modelled by its acoustic impedance $Z_3(f)$ represented by block 93. The sound pressure p_2 at the microphone entrance is converted to an electric signal by the microphone. For the purpose of the present illustration, all [acoustic] elements exposed to the sound pressure, generated by the sound generator SG, except the microphone, are

lumped together in the acoustic impedance $Z_2(f)$ represented by block 95. A leakage in the hearing protector may be modelled by a change in the variable acoustic impedance $Z_2(f)$. The change will usually affect both the frequency dependent modulus and the frequency dependent phase of $Z_2(f)$. This change leads to a change in the relationship between the sound pressures p_2 and p_1 , which is analysed as described in connection with FIG. 8.

Svean, 11:61-13:38.

122. To the extent the Board finds that the “audio device” must be a separate structure from the “earphone,” it is my opinion that *Svean* discloses or suggests this. *Svean* discloses that **electronics unit 11** may be connected to other equipment by a **connection interface 13** that may transmit digital or analog signals. *Id.*, 6:27-31. Further, “**electronic circuitry 11** may be connected to other electrical units by a **bidirectional digital interface E12**” and communicate with those units via a cable or wirelessly. *Id.*, 7:29-37. *Svean* discloses that “signals that may be transmitted through this interface” include, for example, “digitised audio signals in both directions to and from the ear terminal” and “control signals for controlling the operation of the ear terminal.” *Id.*, 7:38-48.



The two microphones and the sound generator are connected to an **electronics unit 11**, which may be connected to other equipment by a **connection interface 13** that may transmit digital or analogue signals, or both, and optionally power.

Svean, 6:27-31.

The **electronic circuitry 11** may be connected to other electrical units by a **bidirectional digital interface E12**. The communication with other electrical units may be performed via a cable or wireless through a digital radio link. The Bluetooth Standard for digital short-range radio Specification of the Bluetooth System, Version 1.0 B, Dec. 1,

1999, Telefonaktiebolaget L M Ericsson) is one possible candidate for wireless communication for this **digital interface E12**.

In a preferred embodiment of the invention, signals that may be transmitted through this interface are:

program code for the **processing unit E3** analysis data from the **processing unit E3** synchronisation data when two ear terminals 1,2 are used in a binaural mode

digitised audio signals in both directions to and from an ear terminal 1,2.

control signals for controlling the operation of the ear terminal.

digital measurement signals for diagnosis of the ear terminal performance.

Svean, 7:29-48.

123. In my opinion, *Svean*'s disclosure that the signals that may be transmitted include "control signals for controlling operation of the ear terminal" contemplates locating a processor and associated memory in a separate device to send such "control signals." Therefore, it is my opinion that a person having ordinary skill in the art would have considered locating the "processor" and "memory" of the "audio device" in a separate structure "configured to test a seal of an earphone" to amount to following *Svean*'s express suggestion to connect the

electronic circuitry of the ear terminal to other electrical units and to transmit “control signals for controlling the operation of the ear terminal” through the interface.

124. Further, in my opinion, this would have amounted to applying a known technique to a known device (locating a processor and memory in a separate structure and communicating control signals from the first device to an ear terminal) ready for improvement to yield predictable results. Indeed, whether the processor and associated memory are located in the ear terminal itself or a separate structure that sends control signals to the ear terminal would predictably achieve the same result—that is, an audio device that tests the seal of an earphone.

125. As to the remaining claim limitations, my analysis is essentially the same unless otherwise noted regardless of whether the “audio device” is the ear terminal or a separate structure. The only difference between these two interpretations is that, under the latter interpretation, the “memory” and “processor” are located in a separate structure from the ear terminal and the processor is configured to execute the instructions and communicate with the ear terminal over **E12/connection interface 13**.

126. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Svean* to disclose or suggest an ear terminal—or a separate device that sends control signals to it—with a memory and a processor 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”

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

128. *Svean* discloses that **processing unit E3** is connected to **RAM (random access memory) E8, ROM (read only memory) E9, and EEPROM (electrically erasable programmable read only memory) E10** (collectively, I will refer to these elements as “**memory elements**”). *Svean*, 7:23-29, Fig. 2. The **memory elements** are “for storing computer programs, filter coefficients, analysis data and other relevant data.” *Id.*, 7:26-29.

129. Fig. 2 shows these **memory elements**.

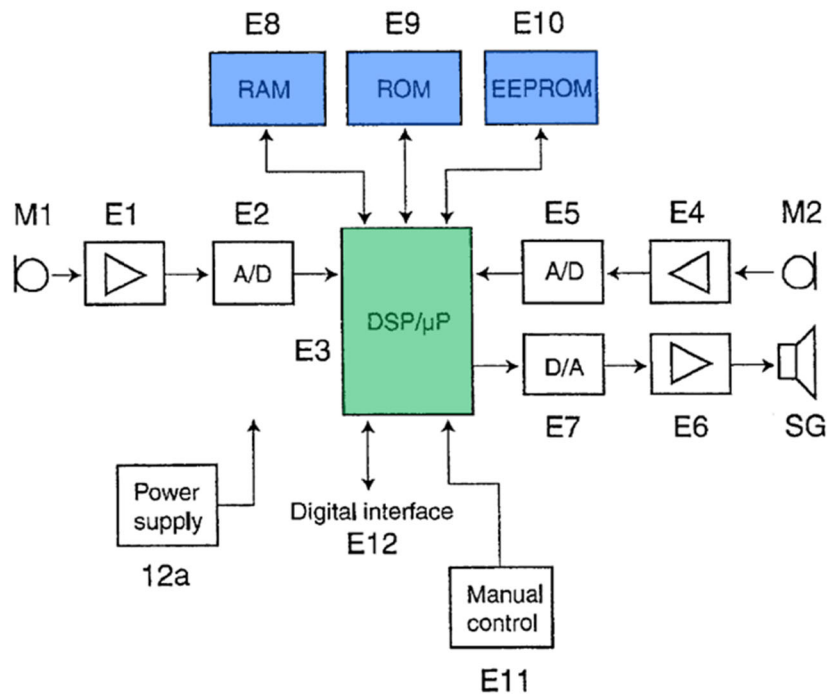


Fig. 2

Svean, Fig. 2.

The **processing unit E3** is connected to **memory elements RAM (Random access memory) E8, ROM (read only memory) E9, and EEPROM (electrically erasable programmable read only memory) E10**. The **memories E8, E9, and E10** are in a preferred embodiment of the invention used for *storing computer programs, filter coefficients, analysis data and other relevant data*.

Id., 7:23-29.

130. In my opinion, a person having ordinary skill in the art would have understood that one of these **memory elements** comprise memory that stores

instructions for the **processing unit E3** to execute. This is because computer programs are a series of instructions that must be stored, and *Svean*'s **processing unit E3** must execute the instructions that are stored in memory to perform the operations that *Svean* describes.

131. To the extent the Board finds that the “memory” must be located in an “audio device” that is a separate structure from the “earphone,” it is my opinion that *Svean* discloses or suggests this as well. *See* §X.A.1.a.

132. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Svean* to disclose or suggest that one of the **memory elements** stores instructions (“memory that stores instructions”) and is located in the ear terminal or in a separate structure.

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

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

134. *Svean* discloses that **processing unit E3** “may be a digital signal processor (DSP), a microprocessor (uP)[,] or a combination of both.” *Svean*, 7:6-10. In addition to **processing unit E3**, *Svean* interchangeably refers to E3 as “digital signal processing (DSP) or microcomputer unit E3” (*id.*, 8:2-5, 8:16-19,

10:58-61, 12:1-6, 12:36-40), “digital signal processor or microprocessor E3” (*id.*, 12:7-10), “DSP/uP processing unit E3” (*id.*, 8:31-33, 9:64-66), “digital signal processor E3” (*id.*, 9:28-29), and “central processing unit E3” (*id.*, 11:26-29). In this declaration, when referring to “**processing unit E3**,” I am referring to each of these alternative terms referring to the E3 component. *Svean* discloses that **processing unit E3** may be located in the **electronics unit 11**. *See, e.g., id.*, Fig. 1, 12:36-41 (“The signal generation and analysis is carried out in a digital signal processing (DSP) or microcomputer unit E3 with appropriate amplifiers and converters as described in the previous paragraphs. All the electronics 11 as well as the power supply 12 are provided in the outer section 1.”). In my opinion, a person having ordinary skill in the art would have understood that **processing unit E3** is a processor, and that it corresponds to the claimed “processor.”

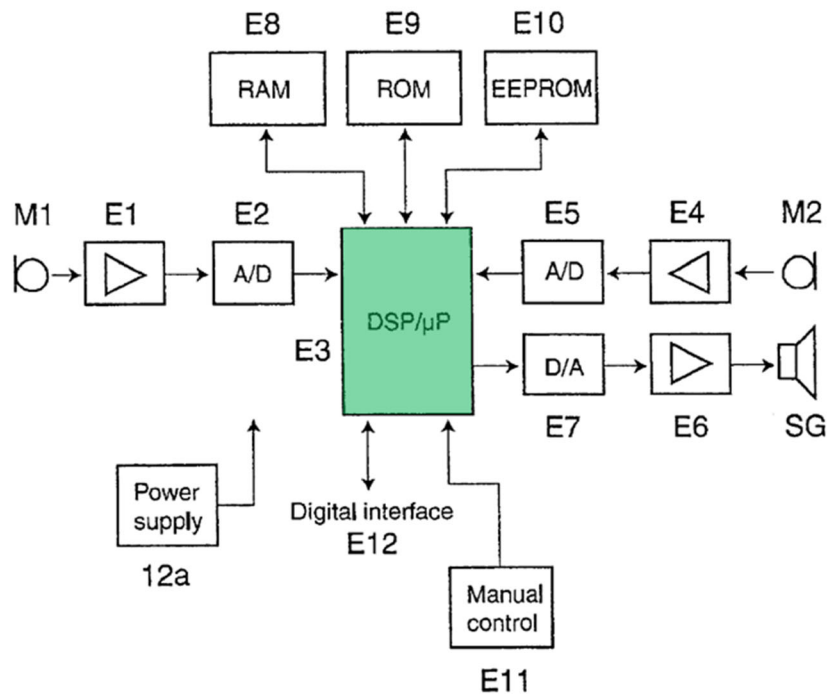


Fig. 2

Svean, Fig. 2.

The **microphone M1** picks up the ambient sound. A signal from the **microphone M1** is amplified in E1 and sampled and digitised in an analogue to digital converter E2 and fed to a **processing unit E3 that may be a digital signal processor (DSP), a microprocessor (uP) or a combination of both**. A signal 51 from **microphone M2**, which picks up the sound in the meatus 3 between the **isolating section 2** and the tympanum 4, is amplified in the amplifier E4 and sampled and digitised in the analogue to digital converter E5 and fed to the **processing unit E3**.

Svean, 7:6-15.

135. *Svean*'s **processing unit E3** executes algorithms ("instructions") to perform various operations including, for example, "signal generation and analysis." *Svean*, 12:36-39; *see also id.*, 12:1-9, 12:42-54; §§ X.A.1.c(i)-X.A.1.c(vi).

The *signal generation and analysis* is carried out in a **digital signal processing (DSP) or microcomputer unit E3** with appropriate amplifiers and converters as described in the previous paragraphs.

Svean, 12:36-40.

A **digital signal processing (DSP) or microcomputer unit E3** in the main section 1 or in the **sealing section 2** is used to *generate a predetermined signal* which is D/A converted by the D/A converter E7, amplified by amplifier E6 and applied to the **sound source SG**, which generates a sound field in the closed part of the meatus 3. The **microphone M2** picks up the sound in the meatus cavity 3. This signal is amplified by amplifier E4, A/D converted by A/D converter E5, and analysed in the **digital signal processor or microprocessor E3**.

Svean, 12:1-9.

FIG. 8 illustrates a signal processing arrangement according to a preferred embodiment of the invention. This embodiment utilises a signal which produces reliable characterisation of the sound field in the cavity, preferably while not being annoying to the user. The signal may comprise one or more sinusoidal components presented

simultaneously, or in sequence. *Alternatively, a pseudorandom sequence may be employed.* In both cases, preferably both the in-phase and the out-of-phase portions of the sound field are analysed and used in the verification algorithm. An example of the signal processing is shown in the flow diagram in FIG. 8. *In the example, two pure tones of different frequencies f_1 and f_2 are generated by algorithms in the processing unit E3.*

Svean, 12:42-54.

136. To the extent the Board finds that the “processor” must be located in an “audio device” that is a separate structure from the “earphone,” it is my opinion that *Svean* discloses or suggests this as well, along with all subsequent limitations relating to the instructions the processor is configured to execute. See §X.A.1.a.

137. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Svean* to disclose or suggest **processing unit E3** (“processor”) located in the ear terminal or a separate structure that is configured to execute the instructions stored in one of the **memory elements** to perform operations such as signal generation and analysis (“configured to execute the instructions to perform operations”).

(i) [1.2.1]: “generating a test signal,”

138. In my opinion, *Svean* discloses or suggests “generating a test signal.”

139. Svean discloses that in the verification embodiment, **processing unit E3** “is used to *generate a predetermined signal*[,] which is D/A converted ..., amplified...[,], and applied to the **sound source SG**, which generates a sound field in the closed part of the meatus 3.” *Id.*, 12:1-6. This sound is picked up by **microphone M2** in the meatus cavity 3. *Id.*, 12:6-7. This signal is amplified, A/D converted, and “analysed in the **digital signal processor or microprocessor E3**.” *Id.*, 12:7-9. “The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions,” and “[t]he user may get audible or other messaged confirmation if the leakage is acceptably low, or a warning signal if leakage is unacceptably high.” *Id.*, 12:9-14; *see also id.* 12:24-54 (disclosing that “*signal generation* and analysis is *carried out*” in the **processing unit E3** and the generated signal “produces reliable characterization of the sound field in the cavity”).

A **digital signal processing (DSP) or microcomputer unit E3** in the main section 1 or in the **sealing section 2** *is used to generate a predetermined signal* which is D/A converted by the D/A converter E7, amplified by amplifier E6 and *applied to the sound source SG, which generates a sound field in the closed part of the meatus 3. The microphone M2 picks up the sound in the meatus cavity 3.* This signal is amplified by amplifier E4, A/D converted by A/D converter E5, and *analysed in the digital signal processor or microprocessor E3*. The result of the analysis is compared to stored results from

previous measurements of the same type in a situation with good sealing conditions. The user may get audible or other messaged confirmation if the leakage is acceptably low, or a warning signal if leakage is unacceptably high.

Id., 12:1-14.

FIG. 1 illustrates an embodiment of the invention where the verification device is integrated in a hearing protective earplug. This embodiment comprises an outer section 1 containing a **microphone M2** and a **sound generator SG**. An inner **sealing section 2** is attached; to the outer section, but may be made in one integrated outer section/sealing section 1,2. An acoustic transmission channel T2 connects **sound generator SG** to the inner portion of the meatus 3. The **sound generator SG** produces a predetermined acoustic signal, which generates a sound field in the meatus 3. An acoustic transmission channel T1 connects **microphone M2** to the inner portion of the meatus 3. **Microphone M2** picks up the sound field being set up by the **sound generator SG**. *The signal generation and analysis is carried out in a digital signal processing (DSP) or microcomputer unit E3* with appropriate amplifiers and converters as described in the previous paragraphs. All the **electronics 11** as well as the power supply 12 are provided in the outer section 1.

FIG. 8 illustrates a signal processing arrangement according to a preferred embodiment of the invention. This embodiment utilises a *signal which produces reliable characterisation of the sound field in the cavity*, preferably while not being annoying to the user. The signal

may comprise one or more sinusoidal components presented simultaneously, or in sequence. *Alternatively, a pseudorandom sequence may be employed.* In both cases, preferably both the in-phase and the out-of-phase portions of the sound field are analysed and used in the verification algorithm. An example of the signal processing is shown in the flow diagram in FIG. 8. In the example, two pure tones of different frequencies f_1 and f_2 are *generated by algorithms in the processing unit E3.*

Svean, 12:24-54.

140. As noted above, in connection with Fig. 8, *Svean* discloses a test signal that comprises two pure tones. *Svean* also discloses an alternative embodiment where its test signal is a “pseudorandom sequence.” *Id.*, 12:42-50. My opinions in this Declaration rely on *Svean*’s “pseudorandom sequence” test signal as corresponding to the claimed “test signal.”

141. Further, I have been asked to confirm whether *Svean*’s disclosure of **processing unit E3’s** generation of a pseudorandom sequence test signal reads on the processor of the ’493 patent’s generation of a test signal, and I confirm that it does. More particularly, the patent discloses, with reference to step 304 of the algorithm depicted in Figure 3, that “[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.

142. Accordingly, in my opinion, a person having ordinary skill in the art would have understood *Svean* to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to generate a pseudorandom sequence test signal (“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;”

143. In my opinion, the *Svean-Voix* combination 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.”

144. In my opinion, a person having ordinary skill in the art would have recognized that *Svean*’s pseudorandom sequence test signal includes multiple frequency components, wherein at least one component lies between a range of 30 Hz and 200 Hz.

145. The most common form of a pseudorandom sequence is an “MLS”/“maximum length sequence”/“m sequence.” Therefore, these terms frequently appear in the prior art, and a person having ordinary skill in the art would have understood them to refer to a pseudorandom sequence. It was well known to a person having ordinary skill in the art as of the priority date of the ’493 patent that MLS sequences, and more generally pseudorandom sequences, include

components of all frequencies in the range they cover and in this sense resemble white noise. This was referred to in the art as having a “constant amplitude” in the frequency domain or a “flat” frequency spectrum. I note that a theoretical “white noise” input has flat spectrum over infinite bandwidth. However, a pseudorandom sequence is considered “white noise” if its bandwidth covers the range of interest.

146. There are numerous prior art references describing these properties of MLS sequences, which a person having ordinary skill in the art would have understood apply to pseudorandom sequences in general. *See, e.g.:*

The MLS signal can be a pseudo random sequence signal of pulses.

In such case, the MLS signal does not have time information. The phase information of the MLS signal may be random and the *amplitude of the MLS signal may be constant across the entire frequency domain.*

EX1012,⁵ 4:58-63.

Since the *pseudorandom sequence has an approximately flat spectrum, it can be considered as a white noise input* if its frequency

⁵ EX1012-EX1016 are cited in this declaration to demonstrate the knowledge of a person having ordinary skill in the art regarding the pseudorandom sequence expressly disclosed in *Svean*. They are not part of the unpatentability grounds in this Declaration.

range is higher than the upper frequency limit of the system under test.

EX1013, 1200.

The magnitude of a pseudo-random noise spectrum in the frequency domain is *basically flat*...

EX1014, 7:7-9.

By emitting through a loudspeaker a *Maximum Length Sequence (resembling a random white noise sequence)* $S(t)$ and measuring by a microphone the Sound pressure $S(t)$ at the desired point, it is able to calculate the transmission path impulse response $h(t)$ *by cross correlation*.

EX1015, [0052].

147. Further, it is my opinion that a person having ordinary skill in the art would have understood that the audible frequency range is 20 Hz - 20 kHz. *See, e.g.:*

A reproduced MLS signal can include a frequency domain in the range of about *20 Hz-20 kHz that is an audible frequency range*.

EX1012, 4:63-65.

The frequency range of human hearing spans from about *20 Hertz (Hz) to 20 kilo-Hertz (kHz)*.

EX1016, 1:8-10.

148. In view of this, it is my opinion that it would have been obvious for *Svean*’s pseudorandom sequence test signal to cover the audible frequency range in view of *Voix*. *Voix* discloses an acoustic seal test for an in-ear device that uses “a ***known noise source signal***, preferably of wide band ranging from ***twenty hertz (20 Hz) to twenty thousand hertz (20000 Hz) for enhanced measurements...***”

EX1020, 7:26-31.

149. In my opinion, a person having ordinary skill in the art would have understood that *Voix*’s disclosures regarding “known noise” are applicable to a pseudorandom sequence because a pseudorandom sequence is a form of “known noise”—i.e., noise that is not truly random and is repeatable.

150. Further, it is my opinion that a person having ordinary skill in the art would have understood that a pseudorandom sequence test signal in the audible range would produce “enhanced measurements” because it would allow the seal test to determine whether there was a good seal either across the entire audible range or for any particular frequency range of interest.

151. As to the latter point, *Voix* discloses that its apparatus may “***provide a difference calculated value for any pre-selectable frequency range*** (any single frequency level included) that is either an average value or a frequency dependent variation over the selected range (such as discrete values for each octave over the frequency range).” *Voix*, 8:56-62. This supports my opinion that a person having

ordinary skill in the art would have recognized the advantages of utilizing a pseudorandom sequence covering the audible frequency range.

152. In view of these advantages, it is my opinion that a person having ordinary skill in the art would have been motivated to utilize a pseudorandom sequence test signal in *Svean* that covers the audible frequency range as *Voix* discloses. Further, it is my opinion that a person having ordinary skill in the art would have had a reasonable expectation of success in implementing a pseudorandom sequence covering the audible frequency range, as this was well-known to a person having ordinary skill in the art. *See, e.g.:*

A reproduced MLS signal can include a frequency domain in the range of about ***20 Hz-20 kHz that is an audible frequency range.***

EX1012, 4:63-65.

153. Accordingly, it is my opinion that a person having ordinary skill in the art would have understood the *Svean-Voix* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to generate a pseudorandom sequence test signal covering the audible range, which is a test signal that includes the claimed (1) “multiple frequency components” (because a pseudorandom sequence has a “flat” frequency spectrum); and (2) “at least one component” that “lies between a range of 30 Hz and 200 Hz.” As to (2), as I explained previously and as the *Voix* and the

background prior art demonstrate was well known to a person having ordinary skill in the art, a pseudorandom sequence in the audible range includes the 20 Hz to 20 kHz range.

- (ii) [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,”

154. In my opinion, the *Svean-Voix* combination 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.”

155. First, *Svean* discloses sending the test signal (*Svean*’s pseudorandom sequence test signal) to a speaker of the ear terminal (**loud speaker/sound generator SG**). *See, e.g., Svean*, 12:1-54 (disclosing that **processing unit E3** “is used to *generate a predetermined signal*[,] which is D/A converted ..., amplified...[,] and applied to the **sound source SG**, which generates a sound field in the closed part of the meatus 3”). As I previously explained, the generated test signal is a pseudorandom sequence. *See* §§X.A.1.c(i) and X.A.1.c(i)(a).

A **digital signal processing (DSP) or microcomputer unit E3** in the main section 1 or in the **sealing section 2** *is used to generate a predetermined signal* which is D/A converted by the D/A converter E7, amplified by amplifier E6 and *applied to the sound source SG, which generates a sound field in the closed part of the meatus 3. The microphone M2 picks up the sound in the meatus cavity 3.* This signal is amplified by amplifier E4, A/D converted by A/D converter E5, and *analysed in the digital signal processor or microprocessor E3.* The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions. The user may get audible or other messaged confirmation if the leakage is acceptably low, or a warning signal if leakage is unacceptably high. In the same manner, a signal may be transmitted to other instances, e.g. an external industrial health monitoring unit, with information about the leakage. One example may be that an ear terminal according to the invention is used for checking for leakage in the hearing protection while the wearer is at a gate controlling admittance to a noise exposed area. If leakage occurs, a signal may be transmitted from the ear terminal to a corresponding signal receiver at the gate, having means to block the gate for entrance until the leakage condition is remedied and verified.

FIG. 1 illustrates an embodiment of the invention where the verification device is integrated in a hearing protective earplug. This embodiment comprises an outer section 1 containing a **microphone M2** and a **sound generator SG**. An inner **sealing section 2** is attached; to the outer section, but may be made in one integrated outer

section/sealing section 1,2. An acoustic transmission channel T2 connects **sound generator SG** to the inner portion of the meatus 3. The **sound generator SG** produces a predetermined acoustic signal, which generates a sound field in the meatus 3. An acoustic transmission channel T1 connects **microphone M2** to the inner portion of the meatus 3. **Microphone M2** picks up the sound field being set up by the **sound generator SG**. *The signal generation and analysis is carried out in a digital signal processing (DSP) or microcomputer unit E3* with appropriate amplifiers and converters as described in the previous paragraphs. All the **electronics 11** as well as the power supply 12 are provided in the outer section 1.

FIG. 8 illustrates a signal processing arrangement according to a preferred embodiment of the invention. This embodiment utilises a *signal which produces reliable characterisation of the sound field in the cavity*, preferably while not being annoying to the user. The signal may comprise one or more sinusoidal components presented simultaneously, or in sequence. *Alternatively, a pseudorandom sequence may be employed*. In both cases, preferably both the in-phase and the out-of-phase portions of the sound field are analysed and used in the verification algorithm. An example of the signal processing is shown in the flow diagram in FIG. 8. In the example, two pure tones of different frequencies f_1 and f_2 are *generated by algorithms in the processing unit E3*.

Svean, 12:1-54.

156. Second, *Svean* discloses that the ear terminal includes the claimed “first microphone,” “speaker,” and “sealing section.” I note that claim 1 does not specify the location of the claimed “first microphone.” However, different dependent claims specify particular locations, depending on the claim. For example, claim 5 recites that the “first microphone” be located on the recited “first side.” On the other hand, claim 6 recites that “first microphone” be located on the recited “second side.” *Svean* discloses microphones located on both the “first side” and the “second side,” and in my opinion both fall within the scope of claims 1’s “first microphone.”

157. In particular, in my opinion, the claimed “first microphone” reads on either of *Svean*’s **microphone (M1)** or **microphone (M2)**. In my opinion, the claimed “speaker” reads on *Svean*’s **speaker (SG)**, while the claimed “sealing section” reads on *Svean*’s **sealing section 2**. See, e.g., *id.*, Fig. 1; *id.*, 11:5-19; *id.*, 11:65-12:1; *id.*, 12:24-30.

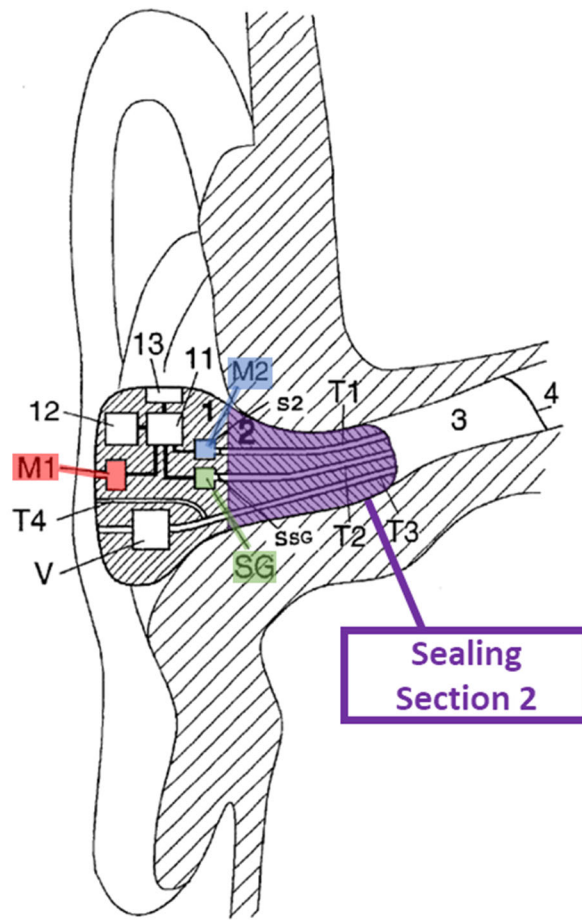


Fig. 1

Svean, Fig. 1.

FIG. 1 shows a preferred embodiment of the invention with the personal noise exposure dose meter integrated in an active hearing protective communications earplug, comprising a main section 1 containing two microphones, an **outer microphone M1** and an **inner microphone M2**, and a **sound generator SG**. Since this embodiment is part of a hearing protecting earplug, a **sealing section 2** is attached to the main section. An acoustic transmission channel T1 connects **microphone M2** to the inner portion of the meatus 3. **Microphone**

M2 therefore picks up the sound present in the meatus 3, just outside the eardrum (tympanum) 4. An acoustic transmission channel T2 connects **sound generator SG** to the inner portion of the meatus 3. The **sound generator SG** may provide audible information to the user, in form of warning signals or synthetic speech.

Svean, 11:5-19.

A small electro-acoustic transducer (**sound source**) **SG** and a **microphone M2** are arranged in a **sealing section 2** arranged for attenuating sounds entering the meatus cavity 3.

Svean, 11:65-12:1.

FIG. 1 illustrates an embodiment of the invention where the verification device is integrated in a hearing protective earplug. This embodiment comprises an outer section 1 containing a **microphone M2** and a **sound generator SG**. An inner **sealing section 2** is attached; to the outer section, but may be made in one integrated outer section/sealing section 1,2.

Svean, 12:24-30.

158. Third, *Svean* discloses that **sealing section 2** is configured to form an acoustic barrier between a **first side** of the earphone (the side adjacent to the meatus of ear canal 3) and a **second side** of the earphone (the side adjacent to the ambient environment) when the earphone is worn by a user. *See, e.g., id.*, Fig. 1; *id.*, 11:5-19; *id.*, 11:65-12:1; *id.*, 12:24-30; *id.*, 5:4-8; *id.*, Claim 1 (“1. Ear

protecting device comprising: a **sealing section (2)** *for acoustically sealing* the meatus (3) of a human...”).

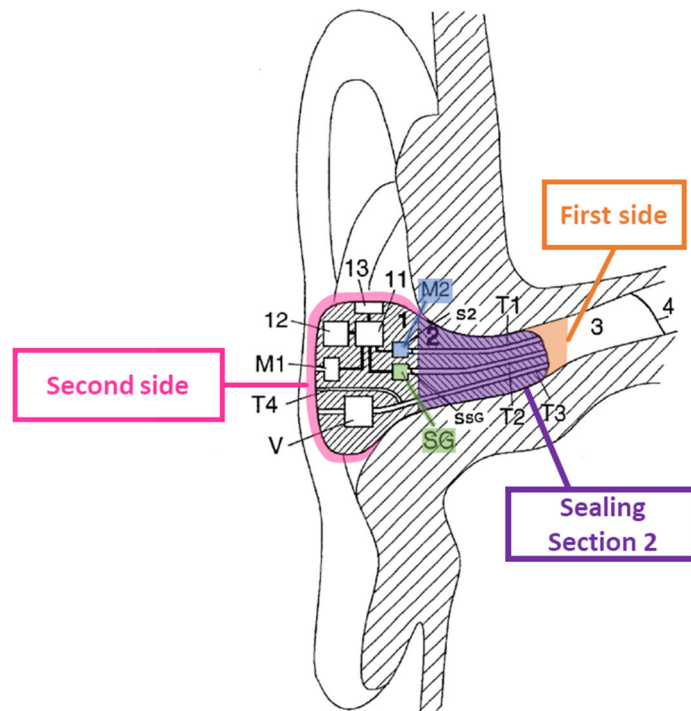


Fig. 1

Svean, Fig. 1.

FIG. 1 shows a preferred embodiment of the invention with the personal noise exposure dose meter integrated in an active hearing protective communications earplug, comprising a main section 1 containing two microphones, an **outer microphone M1** and an **inner microphone M2**, and a **sound generator SG**. Since this embodiment is part of a hearing protecting earplug, a **sealing section 2** is attached to the main section. An acoustic transmission channel T1 connects **microphone M2** to the inner portion of the meatus 3. **Microphone**

M2 therefore picks up the sound present in the meatus 3, just outside the eardrum (tympanum) 4. An acoustic transmission channel T2 connects **sound generator SG** to the inner portion of the meatus 3. The **sound generator SG** may provide audible information to the user, in form of warning signals or synthetic speech.

Svean, 11:5-19.

A small electro-acoustic transducer (**sound source**) **SG** and a **microphone M2** are arranged in a **sealing section 2** arranged for attenuating sounds entering the meatus cavity 3.

Svean, 11:65-12:1.

FIG. 1 illustrates an embodiment of the invention where the verification device is integrated in a hearing protective earplug. This embodiment comprises an outer section 1 containing a **microphone M2** and a **sound generator SG**. An inner **sealing section 2** is attached; to the outer section, but may be made in one integrated outer section/sealing section 1,2.

Svean, 12:24-30.

The ear terminal comprises an outer section 1 arranged for sitting adjacent to the outward facing portion of the **sealing section 2** and a part of the inward facing portion of the outer section 1 is formed to fit the concha around the outer portion of the meatus 3.

Svean, 5:4-8.

1. Ear protecting device comprising:

a **sealing section (2)** for *acoustically sealing* the meatus (3) of a human.

Svean, Claim 1.

159. In addition, I have been asked to confirm whether Svean's disclosure of **processing unit E3's** sending of a pseudorandom sequence testing signal to **loudspeaker/sound generator SG** reads on the '493 patent's disclosure of a processor that sends a test signal to a speaker of an earphone, and I confirm that it does. More particularly, the '493 patent discloses, in connection with step 304 of the algorithm depicted in Figure 3, that "[t]he test signal is provided to the *acoustic transducer*," which, the patent discloses, is a "loudspeaker." '493 patent, 8:8-10.

160. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to send the pseudorandom sequence test signal to **loudspeaker/sound generator SG** of the ear terminal ("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 the *Svean-Voix* combination to disclose or suggest that the ear terminal includes *Svean's* **microphone M1** and **microphone M2** (either of which corresponding to the "first microphone"), **loud speaker/sound generator SG** ("speaker"), and **sealing section 2** ("sealing section"). Finally, in

my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix* combination to disclose or suggest that *Svean*'s ear terminal includes a **sealing section 2** that attenuates sounds between the ear canal 3 on the **first side** and the ambient environment on the **second side** of the ear terminal (“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) [1.2.3]: “receiving a first microphone signal from the first microphone;”

161. In my opinion, the *Svean-Voix* combination discloses or suggests “receiving a first microphone signal from the first microphone.”

162. Where *Svean*'s **microphone M2** is the claimed “first microphone,” *Svean* discloses that **processing unit E3** receives a first microphone signal from **microphone M2** in *Svean*'s “verification embodiment.” *Svean*, 12:1-9; *id.*, 5:66-6:1; *id.*, 6:12-14; *id.*, 7:10-15; *id.*, Fig. 2.

A **digital signal processing (DSP) or microcomputer unit E3** in the main section 1 or in the **sealing section 2** *is used to generate a predetermined signal* which is D/A converted by the D/A converter E7, amplified by amplifier E6 and *applied to the sound source SG, which generates a sound field in the closed part of the meatus 3. The microphone M2 picks up the sound in the meatus cavity 3.* This signal is amplified by amplifier E4, A/D converted by A/D converter

E5, and *analysed in the digital signal processor or microprocessor E3*.

Id., 12:1-9.

FIG. 1 illustrates an embodiment according to the invention. The earplug comprises a main section 1 containing two microphones **M1** and **M2** and a **sound generator SG**.

Svean, 5:66-6:1.

The **microphone M2** is connected to the inner portion of the meatus 3 by means of an acoustic transmission channel T1.

Svean, 6:12-14.

A signal 51 from **microphone M2**, which picks up the sound in the meatus 3 between the isolating section 2 and the tympanum 4, is amplified in the amplifier E4 and sampled and digitised in the analogue to digital converter E5 and fed to the **processing unit E3**.

Svean, 7:10-15.

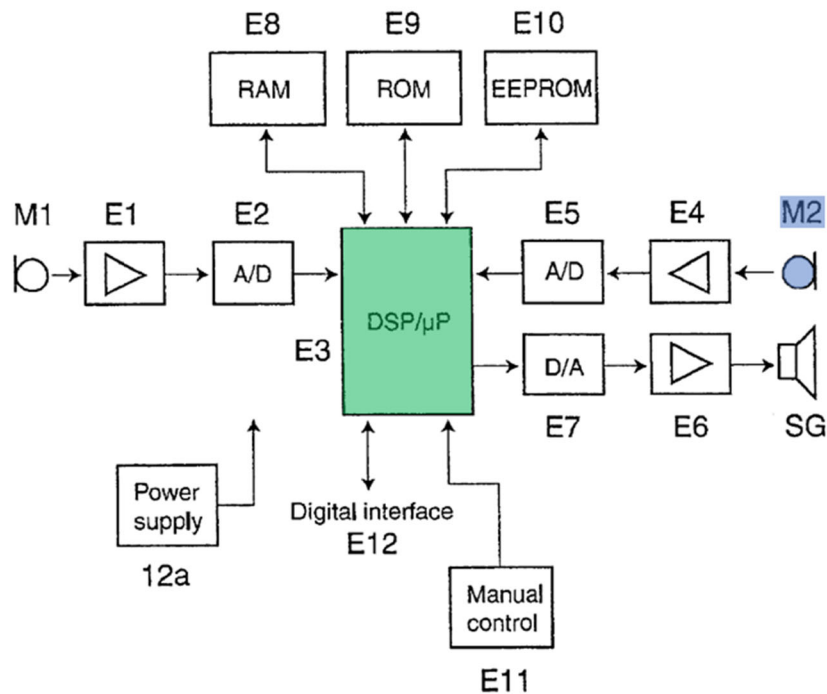


Fig. 2

Svean, Fig. 2.

163. Where *Svean*’s **microphone M1** is the claimed “first microphone,” *Svean* discloses that **processing unit E3** receives a first microphone signal from **microphone M1**. *Id.*, 7:6-10; *id.*, 5:66-6:1; *id.*, 6:10-12; *id.*, Fig. 2.

The **microphone M1** picks up the ambient sound. A signal from the **microphone M1** is amplified in E1 and sampled and digitised in an analogue to digital converter E2 and *fed to a processing unit E3* that may be a digital signal processor (DSP), a microprocessor (uP) or a combination of both.

Svean, 7:6-10.

FIG. 1 illustrates an embodiment according to the invention. The earplug comprises a main section 1 containing two microphones **M1** and **M2** and a **sound generator SG**.

Svean, 5:66-6:1.

The sound inlet of **microphone M1** is connected to the outside of the earplug, picking up the external sounds.

Svean, 6:10-12.

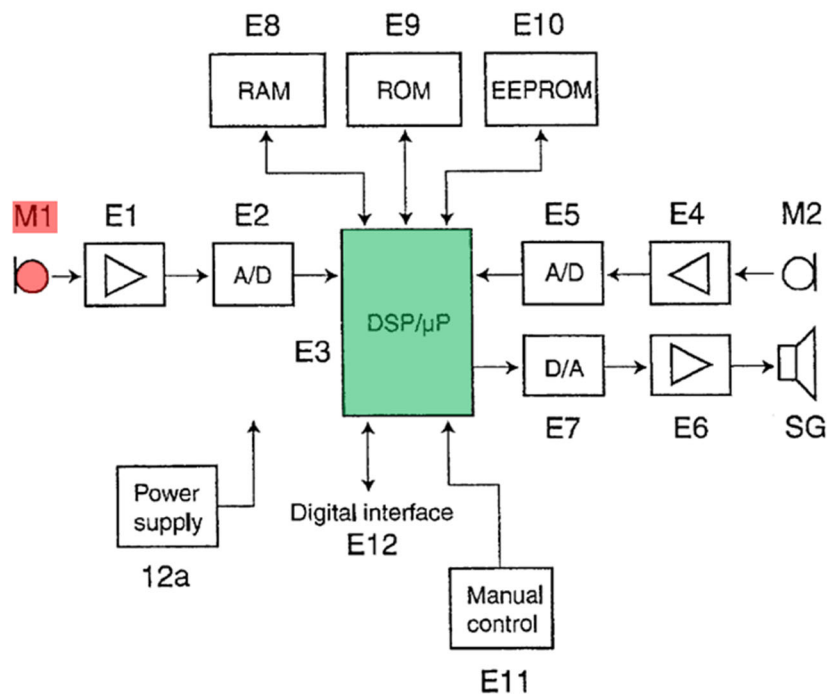


Fig. 2

Svean, Fig. 2.

164. Although *Svean* does not explicitly disclose that **processing unit E3** receives a microphone signal from **microphone M1** in the “verification”

embodiment (*see id.*, 11:61-13:38), it is my opinion that a person having ordinary skill in the art would have understood and found it obvious that *Svean* would use **microphone M1** to collect external sounds—and **processing unit E3** would be configured to receive the collected sounds—while performing “verification.” This is because it is my opinion that a person having ordinary skill in the art would have understood and found it obvious that external sounds should be provided to a user while verifying the device, and *Svean* discloses that **microphone M1** is involved in collecting such “external sounds.” *Id.*, 5:24-41. Indeed, *Svean* contemplates that the “verification” process may be run for long time periods—even continuously—such that receiving and providing external sounds to the user during “verification” would be necessary. *Id.*, 13:13-20; *id.*, 4:6-7.

Restoration of desired sounds (external sounds and signals from the communication system) at the eardrum or tympanum 4 is achieved by using the same microphones **M1**, **M2**, and **loudspeaker SG** and the **electronics unit 11**. Again, the algoritimes [sic] for obtaining this are per se known and will-not be described in any detail here, but may include amplification of chosen frequencies converted by said the **microphone (M1)** and generating a corresponding acoustic signal through said **sound generator (SG)**. The frequencies may for example be within the normal range of the human voice. Ear terminal according to claim 1, comprising a **sound generator (SG)** arranged for being directed toward the meatus and being coupled to said

electronics unit (11), wherein the **electronics unit (11)** comprises filtering means for active sound transmission e.g. by amplification of chosen frequencies converted by said **outer microphone (M1)** and generating a corresponding acoustic signal through said **sound generator (SG)**.

Svean, 5:24-41.

The number and values of frequencies and the Smoothing characteristics of the detectors are chosen as a compromise between audibility and response time. ***If a continuously running verification should be necessary***, low frequencies, e.g. in the range of 10-20 Hz, of sufficiently low levels may be utilised in order to avoid annoyance. The pure tones may then be partially or fully aurally masked by the residual noise transmitted by the hearing protector.

Svean, 13:13-20.

Verification may be activated by the user at any time, or be ***continuously running*** when the application is critical.

Svean, 4:6-7.

165. In addition, I have been asked to confirm whether **processing unit E3's** receiving a signal from **microphone M2** or from **microphone M1** reads on the '493 patent's disclosure of a processor that receives a microphone signal from a microphone, and I confirm that it does. More particularly the '493 patent discloses, with respect to step 308 of the algorithm depicted in Figure 3, that "[t]he acoustic

field in the first volume is detected by an Ear Canal Microphone (ECM) in a step 308.” ’493 patent, 8:12-13. Moreover, the patent discloses that “ECM 106 is configured to measure the test signal in ear canal 124 and *provide the measured test signal to processor 116.*” *Id.*, 8:58-60.

166. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** for receiving a microphone signal from **microphone M1** or **microphone M2** (“receiving a first microphone signal from the first microphone.”).

- (iv) **[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;”**

167. In my opinion, the *Svean-Voix-Padhi* combination 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.”

168. 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, *Svean* discloses calculating a “value” using a “first signal” and a “second signal.”

169. When *Svean*’s test signal comprises two pure tones of frequencies f_1 and f_2 , *Svean* discloses blocks 84 through 87 that perform “a sample by sample multiplication of [] two input signals” and “smooth[] the result with a low pass filter.” *Svean*, 12:51-66; *id.*, Fig. 8.

An example of the signal processing is shown in the flow diagram in FIG. 8. In the example, two pure tones of different frequencies f_1 and f_2 are generated by algorithms in the **processing unit E3**. The generators are represented by blocks 81 and 82 respectively. The generators generate both the in-phase (sin) and out-of-phase (cos) components. The in-phase components are added together in block 83, converted to analogue form, amplified and applied to the **sound generator SG**. The resulting sound field is picked up by the **microphone M2**, amplified, converted to digital form and analysed by algorithms in the **processing unit E3** for a series of detectors represented by blocks 84, 85, 86 and 87. The in-phase and out-of-phase components of the **microphone M2 signal** are analysed for each of the two frequencies. The detector algorithm performs a sample by sample multiplication of the two input signals and smoothes the result with a low-pass filter.

Svean, 12:51-66.

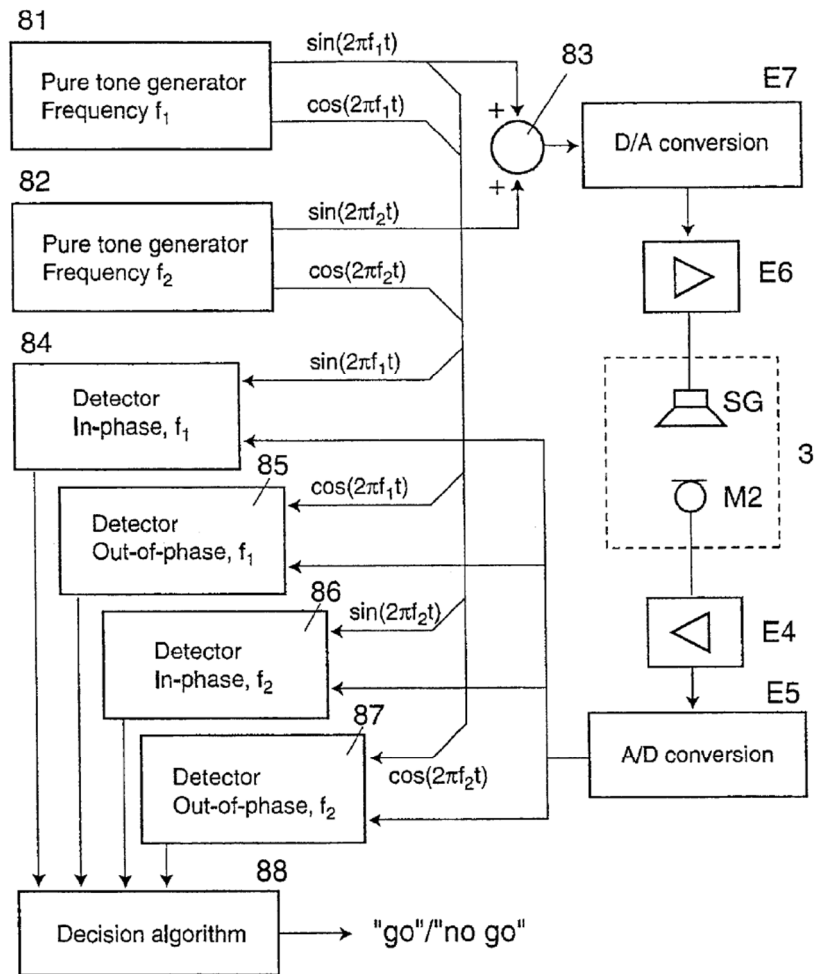


Fig. 8

Svean, Fig. 8.

170. *Svean* calculates a value using these test signal components and a signal output from **microphone M2**, which is also input to blocks 84-87. The blocks' outputs "are applied to a decision algorithm represented by block 88 where they are compared to stored values." *Id.*, 12:66-13:11.

The four detector outputs ***are applied to a decision algorithm represented by block 88 where they are compared to stored values.***

The decision result may be a digital “go”/“no go” real time signal indicating acceptable noise protection attenuation or unacceptable protection conditions. The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions. The stored values for the decision algorithm may according to a preferred embodiment be based on previous laboratory experiments, but values for the decision algorithm may also be determined, e.g. making an average and setting a lower acceptance limit for a general-purpose embodiment of the invention.

Svean, 12:66-13:11.

171. In my opinion, a person having ordinary skill in the art would have understood that because the outputs “are compared to stored values,” the outputs are themselves calculated values. In other words, it would not be possible to compare the outputs to stored values unless the outputs are themselves values. Further, in my opinion, a person having ordinary skill in the art would have understood that the outputs of blocks 84-87 indicate the similarity between the two input signals because they are values calculated by multiplying the two input signals. A person having ordinary skill in the art would have understood that the result of the multiplication would indicate the similarity of the signals—for example, a multiplication that results in a positive value that is relatively large in magnitude would indicate that the two input signals are similar.

172. However, *Svean*'s Figure 8 algorithm is not designed for calculating such a value when the test signal is a pseudorandom sequence. Although *Svean* discloses that “preferably⁶ [sic] both the in-phase and the out-of-phase portions of the sound field are analysed [sic] and used in the verification algorithm” (*Svean*, 12:48-50), it is my opinion that a person having ordinary skill in the art would have recognized that it would be extremely computationally intensive to implement an algorithm that requires calculations with sines and cosines of each frequency for a pseudorandom sequence, which—unlike the two pure tones analyzed in Fig. 8—has a “flat” frequency spectrum with signal content at essentially every frequency within the range. For example, the number of frequencies present in a pseudorandom sequence could approach the thousands or even millions, and it would not be practical to calculate the sines and cosines at each of these frequencies.

⁶ In my opinion, this disclosure in *Svean* also does not criticize, discredit, or otherwise discourage an alternative verification algorithm such as *Padhi*'s cross correlation. Thus, I understand that *Svean* does not “teach away” from using *Padhi*'s cross correlation algorithm.

173. Thus, when *Svean*'s "verification" is performed with *Svean*'s pseudorandom sequence test signal, it is my opinion that a person having ordinary skill in the art would have been motivated to look to other teachings for implementation details for calculating such a value and would have readily identified *Padhi*.

174. *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. It was well known to a person having ordinary skill in the art that cross correlation was suitable for calculating values with pseudorandom sequence signals. *See, e.g.:*

By emitting through a loudspeaker a ***Maximum Length Sequence (resembling a random white noise sequence)*** $S(t)$ and measuring by a microphone the Sound pressure $S(t)$ at the desired point, it is able to calculate the transmission path impulse response $h(t)$ ***by cross correlation.***

EX1015, [0052].

The output of the microphone 20 is fed to a computer 30. As noted, the computer 30 controls the MLS noise generator. It also performs ***cross-correlation of the original MLS noise and the output of the microphone***, providing the impulse response of the total System.

EX1018, 6:34-38.

If an MLS is applied to a DLTI system with [periodic impulse response] $h'[n]$, the system output $y'[n]$ is

$$y'[n] = s'[n] \star h'[n] = \sum_{k=0}^{L-1} s'[k]h'[n - k] . \quad (8)$$

To compute the [periodic impulse response] for the system, the system output and input are *cross-correlated*.

EX1019, 585.

175. *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].

176. In the *Svean-Voix-Padhi* combination, *Svean*'s **microphone M2** signal is supplied as **input audio signal 102** and *Svean*'s pseudorandom sequence test signal is supplied as **reference signal 104**.

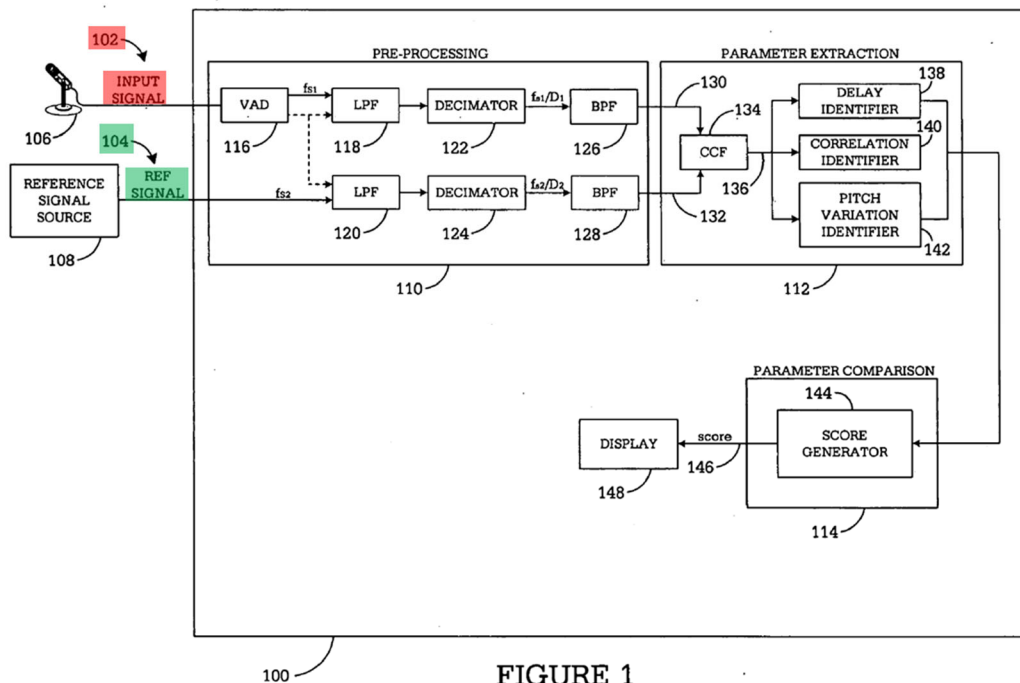


FIGURE 1

Padhi, Fig. 1.

177. 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** (*Svean*’s **M2 signal**) and the **reference signal 104** (*Svean*’s pseudorandom sequence test signal). *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].

178. At [0077]-[0082], *Padhi* describes how score generator 144 generates a score. “First, the score generator 144 generates a value α using information from the delay identifier 138 and the correlation identifier 140.” *Id.*, [0077]. The value α 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 α using the information from the delay identifier 138 and the correlation identifier 140. The value of α 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 (± 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 α , the score generator 144 selects a first score (Score₁) using Table 1.

Padhi, [0078].

179. In my opinion, *Padhi*'s α is a "value" that was calculated "using a first signal and a second signal" (*Svean*'s **microphone M2** signal and the pseudorandom test signal), "wherein the value is the result of a non phase calculation" that is a "cross-correlation." In my opinion, this is sufficient for this limitation, and a person having ordinary skill in the art would have recognized that *Padhi*'s additional description regarding pitch (and values β_1 and β_2) would be unnecessary for cross correlating two audio signals for a seal test. *See Padhi*, [0079]-[0080].

After that, the score generator 144 uses the information from the pitch variation identifier 142 to select a second score, which is added to the first score to produce the final score 146. In particular, the score generator 144 produces two values β_1 and β_2 , which uses the estimated coincidental harmonic frequency for three successive frames 202 calculated using Equation 5. The score generator 144 uses the following equations to identify values for β_1 and β_2 :

$$\beta_1 = \left| \frac{Pitch_{est}^{l-1} - Pitch_{est}^{l-2}}{Pitch_{est}^{l-2}} \right| \quad (7)$$

$$\beta_2 = \left| \frac{Pitch_{est}^l - Pitch_{est}^{l-1}}{Pitch_{est}^{l-1}} \right|. \quad (8)$$

Padhi, [0079].

Ideally, if the **input signal 102** has a constant pitch, the deviation of the estimated coincidental harmonic frequency across the frames 202 should be small. Using the values of β_1 and β_2 , the score generator 144 assigns a final score using Table 2. In Table 2, if both the “Range 1” condition and “Range 2” condition in the same line are satisfied, the final score is computed using the formula on that line. The formula adds the first score to a second score and produces the final score 146. In Table 2, the symbol $\vee$ denotes the logical OR operation.

Padhi, [0080].

180. In my opinion, a person having ordinary skill in the art would not only have been motivated to look to *Padhi* for implementation detail, but also would have found *Padhi*'s technique to be an efficient and flexible solution that provides a calculated value that can then be compared to a threshold—just as *Svean*'s Figure 8 algorithm compares to stored values (i.e., thresholds) for a test signal that comprises two pure tones. Indeed, *Padhi* discloses that the reference and microphone signal that are inputs to its calculation “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].

181. In my opinion, a person having ordinary skill in the art would also have considered the combination of *Svean* and *Voix* with *Padhi* to amount to the use of a known technique (calculating a value with two signals using cross correlation) to improve a similar device (*Svean*'s verification algorithm that involves calculating a value and comparing to thresholds to determine the seal) with a predictable result. Indeed, I note that the Examiner in the parent '836 patent *ex parte* reexamination 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." EX1017, 27. In my opinion, the same reasoning applies to the '493 patent, because *Padhi*'s cross correlation compares two signals by calculating a value with the two signals.

182. In my opinion, a person having ordinary skill in the art would also have had a reasonable expectation of success in combining *Svean*, *Voix*, and *Padhi*. As previously discussed, it was well known to a person having ordinary skill in the art that cross correlation was appropriate for pseudorandom sequences such as MLS signals. *See, e.g.*, EX1015, [0052] (disclosing that calculating the impulse response of a maximum length sequence emitted through a loudspeaker and measured by a microphone "***by cross correlation***"); EX1018, 6:34-38 (disclosing "***cross-correlation of the original MLS noise and the output of [a] microphone,***

providing the impulse response of the total system”); EX1019, 585 (describing testing a hearing aid with a maximum length sequence (MLS) and explaining that “[t]o compute the [periodic impulse response] for the system, the system output and input are *cross-correlated*”).

183. In addition, I have been asked to confirm whether **processing unit E3**’s calculation of a cross-correlation using the pseudorandom test signal and the **microphone M2 signal** as described above reads on the ’493 patent’s disclosure of a processor that calculates a cross-correlation of a test signal and microphone signal, and I confirm that it does. More particularly the ’493 patent discloses, with respect to step 310 of the algorithm depicted in Figure 3, that “[t]he 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.*” ’493 patent, 8:26-29.

184. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to calculate a value that is the result of a non-phase cross correlation calculation (α) with the pseudorandom sequence test signal and the **microphone M2 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) **[1.2.5]: “comparing the value to a threshold to determine a seal quality between the first side and the second side;”**

185. In my opinion, the *Svean-Voix-Padhi* combination teaches or suggests “comparing the value to a threshold to determine a seal quality between the first side and the second side.”

186. In my opinion, *Padhi*’s algorithm also discloses comparing a calculated value to a threshold. With respect to α , *Padhi* discloses comparing this value to a threshold to assign Score_1 —for example, if α is greater than or equal to .75, Score_1 is assigned the highest value (60). *Padhi*, [0078]. See Table 1:

TABLE 1

Range	Score_1
$\alpha \geq 0.75$	60
$0.75 > \alpha \geq 0.5$	50
$0.5 > \alpha \geq 0.25$	40
$0.25 > \alpha$	30

Padhi, Table 1.

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 (± 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 α , the score generator 144 selects a first score (Score₁) using Table 1.***

Padhi, [0078].

187. In my opinion, in the *Svean-Voix-Padhi* combination, comparison to a threshold would demonstrate the level of similarity between the test signal and the **microphone M2 signal**. 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 *Svean*, *Voix*, and *Padhi* 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.

188. In my opinion, a person having ordinary skill in the art would have been motivated to combine *Svean*, *Voix*, and *Padhi* for the same reasons I described with respect to [1.2.4]. See §X.A.1.c(iv). As I described with respect to [1.2.4], *Svean* discloses that its detector outputs are compared to stored results (i.e., thresholds). *Svean* produces a decision result that is a “go”/“no go” real time signal. *Svean*, 12:66-13:11.

The four detector outputs ***are applied to a decision algorithm represented by block 88 where they are compared to stored values.***
The decision result may be a digital “go”/“no go” real time signal

indicating acceptable noise protection attenuation or unacceptable protection conditions. The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions. The stored values for the decision algorithm may according to a preferred embodiment be based on previous laboratory experiments, but values for the decision algorithm may also be determined, e.g. making an average and setting a lower acceptance limit for a general-purpose embodiment of the invention.

Svean, 12:66-13:11.

189. Thus, in the combination, it is my opinion that a person having ordinary skill in the art would have understood that *Padhi*'s α would similarly be compared to a threshold (e.g., 0.75) to produce a “go”/“no go” signal just as *Svean* discloses, which is a determination of “seal quality” to the extent the Board finds that seal quality may be a binary result because it indicates whether the seal is good or bad.

190. In my opinion, a person having ordinary skill in the art also would have been motivated to use *Padhi*'s calculation, comparison to a threshold, and resulting score because a person having ordinary skill in the art would have recognized that this would have the advantage of providing the user more useful and granular information in assessing the “seal quality” than a binary result. For example, a person having ordinary skill in the art would have understood that

providing a score indicates how far above or below the threshold the seal quality is, which allows the user to make a decision as to whether to attempt to reposition the ear terminal and perform another seal test to see if the seal quality—even if adequate—can be improved.

191. In addition, I have been asked to confirm whether **processing unit E3's** comparing of the calculated value α to a threshold (*e.g.*, 0.75) to produce *Svean's* “go”/“no go,” as I described above, reads on the '493 patent's disclosure of a processor that compares a calculated value to a threshold, and I confirm that it does. More particularly the '493 patent discloses, with respect to steps 416-420 of the algorithm depicted in Figure 4, that

[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*. 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, 9:45-52.

192. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to compare a calculated value (α) to a threshold (comparing α

to, e.g., .75) and to determine the seal quality (i.e., *Svean*'s "go"/"no go" real time signal or *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) [1.2.6] and [1.2.7]: "generating a message to convey the seal quality; and sending the message to the user."

193. In my opinion, The *Svean-Voix-Padhi* combination discloses or suggests "generating a message to convey the seal quality; and sending the message to the user."

194. To the extent the Board finds that seal quality may be a binary result (e.g., "good" or "bad"), *Svean* discloses or suggests generating a message that conveys the "go"/"no go" decision and sending the message to the user. In particular, *Svean* discloses that "the user may get audible or other messaged confirmation if the leakage is acceptably low, or a warning signal if leakage is unacceptably high." *Svean*, 12:6-14; *id.*, 12:66-13:11.

The **microphone M2** picks up the sound in the meatus cavity 3. This signal is amplified by amplifier E4, A/D converted by A/D converter E5, and analysed in the **digital signal processor or microprocessor E3**. The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions. *The user may get audible or other messaged*

confirmation if the leakage is acceptably low, or a warning Signal if leakage is unacceptably high.

Svean, 12:6-14.

The four detector outputs *are applied to a decision algorithm represented by block 88 where they are compared to stored values. The decision result may be a digital “go”/“no go” real time signal indicating acceptable noise protection attenuation or unacceptable protection conditions.* The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions. The stored values for the decision algorithm may according to a preferred embodiment be based on previous laboratory experiments, but values for the decision algorithm may also be determined, e.g. making an average and setting a lower acceptance limit for a general-purpose embodiment of the invention.

Svean, 12:66-13:11.

195. Otherwise, in my opinion, *Padhi* discloses or suggests these features.

Padhi discloses generating a score that “is presented to a user or other person or people on a **display 148**.” *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].

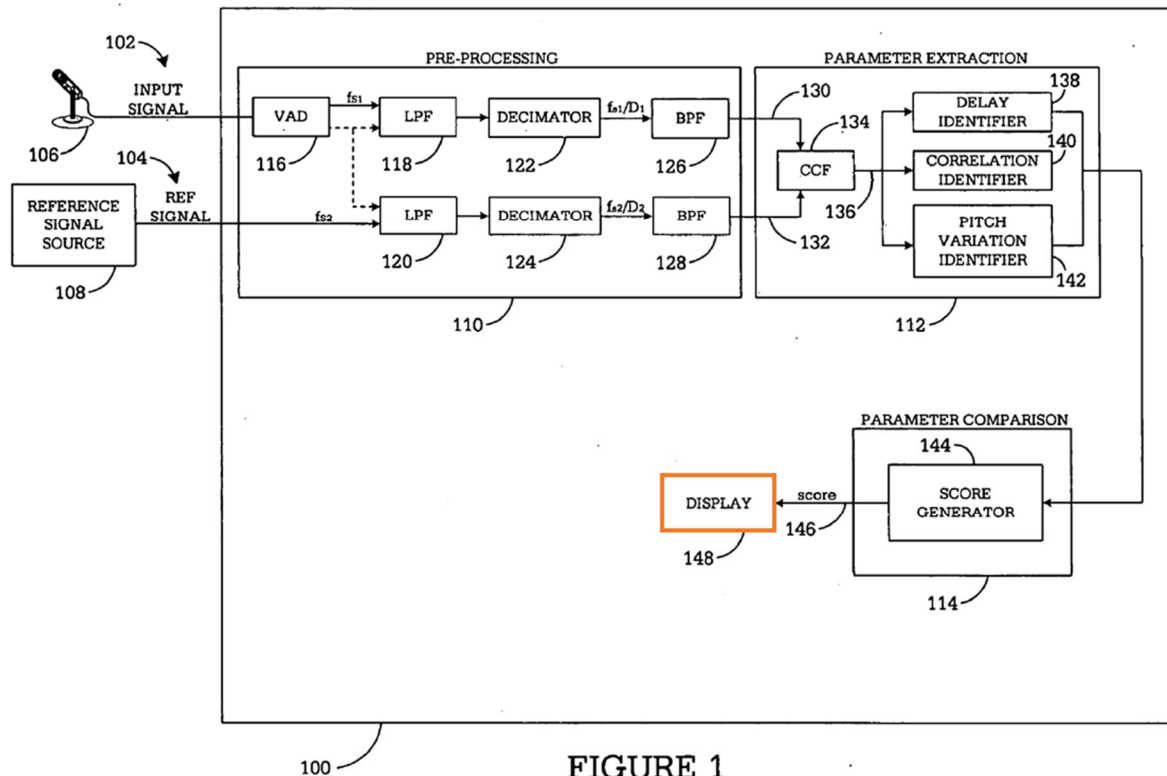


FIGURE 1

Padhi, Fig. 1.

196. *Padhi*'s score not only conveys a binary result, but also conveys granular information to the user regarding the seal level. In my opinion, a person having ordinary skill in the art would have been motivated to generate and send a message to the user that conveys this more granular information for the reasons discussed in §X.A.1.c(v).

197. In addition, I have been asked to confirm whether **processing unit E3's** providing of an audible “go”/“no go” decision and its display (such as via a **display 148**) of a score that conveys seal quality,” as I described above, reads on the '493 patent's disclosure of a processor that generates and sends to a user a message that conveys seal quality, and I confirm that it does. More particularly the '493 patent discloses, with respect to steps 418 and 420 of the algorithm depicted in Figure 4, that, “[i]f 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. *See also id.*, 15:63-16:2 (“the degree of acoustic seal integrity is expressed as either a PASS or FAIL status ... [A]n earpiece that has performed the seal test and determined that the sealing unit is not sealed correctly ... can provide a signal or message indicating the problem.”).

198. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to generate and send a message to the user that conveys either *Svean's* binary result or *Padhi's* score regarding seal quality to the user

(“generating a message to convey the seal quality” and “sending the message to the user”).

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

199. In my opinion, the *Svean-Voix-Padhi* combination discloses or suggests “[t]he audio device according to claim 1, wherein the message is an acoustic message.”

200. In my opinion, to the extent the Board finds that seal quality may be a binary result, *Svean*’s “audible or other messaged confirmation” is an acoustic message. *Svean*, 12:6-14; *id.*, 12:66-13:11.

The **microphone M2** picks up the sound in the meatus cavity 3. This signal is amplified by amplifier E4, A/D converted by A/D converter E5, and analysed in the **digital signal processor or microprocessor E3**. The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions. *The user may get audible or other messaged confirmation if the leakage is acceptably low, or a warning Signal if leakage is unacceptably high.*

Svean, 12:6-14.

The four detector outputs *are applied to a decision algorithm represented by block 88 where they are compared to stored values. The decision result may be a digital “go”/“no go” real time signal indicating acceptable noise protection attenuation or unacceptable*

protection conditions. The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions. The stored values for the decision algorithm may according to a preferred embodiment be based on previous laboratory experiments, but values for the decision algorithm may also be determined, e.g. making an average and setting a lower acceptance limit for a general-purpose embodiment of the invention.

Svean, 12:66-13:11.

201. Otherwise, it is my opinion that *Padhi* discloses or suggests these features because *Padhi* teaches that it is desirable for such messaging to convey more granular seal quality information (a score). *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].

202. In my opinion, a person having ordinary skill in the art implementing the *Svean-Voix-Padhi* combination would have been motivated to provide *Padhi*'s seal quality score in an audible format, as in *Svean*, because as in *Svean*, providing

an audible message would be another method of ensuring that the user is aware of seal quality information.

203. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to generate and send an audible message to the user that conveys either a binary result or a score regarding seal quality to the user (“acoustic message”).

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

204. In my opinion, the *Svean-Voix-Padhi* combination discloses or suggests “[t]he audio device according to claim 1, wherein the speaker plays the test signal from the first side.”

205. This limitation requires that the speaker is located on the **first side** (i.e., inside the ear canal) and plays the test signal from this location.

206. *Svean* discloses that it is “possible to mount ... the **sound generator SG** at the innermost part of the sealing section.” *Svean*, 6:22-26.

When smaller **microphones M2**, and **sound generators SG** are available, *it will be possible to mount the microphone M2 and the sound generator SG at the innermost part of the sealing section.*

Then there is no need for the transmission channels T1 and T2.

Svean, 6:22-26.

207. In that case, it is my opinion that **SG** is a “speaker” located on the **first side** of the earphone and “plays the test signal from” that location.

208. In addition, *Svean* discloses in Fig. 1 that speaker **SG** is connected through a **transmission channel T2** to the inner portion of the meatus 3 so that the speaker plays a test signal propagating from the **first side**. *Svean*, 12:30-31; *id.*, 11:65-12:6.

An acoustic **transmission channel T2** connects **sound generator SG** to the inner portion of the meatus 3.

Svean, 12:30-31.

A small **electro-acoustic transducer (sound source) SG** and a **microphone M2** are arranged in a **sealing section 2** arranged for attenuating sounds entering the meatus cavity 3. A **digital signal processing (DSP) or microcomputer unit E3** in the main section 1 or in the **sealing section 2** is used to generate a predetermined signal which is D/A converted by the D/A converter E7, amplified by amplifier E6 and applied to the **sound source SG**, *which generates a sound field in the closed part of the meatus 3*.

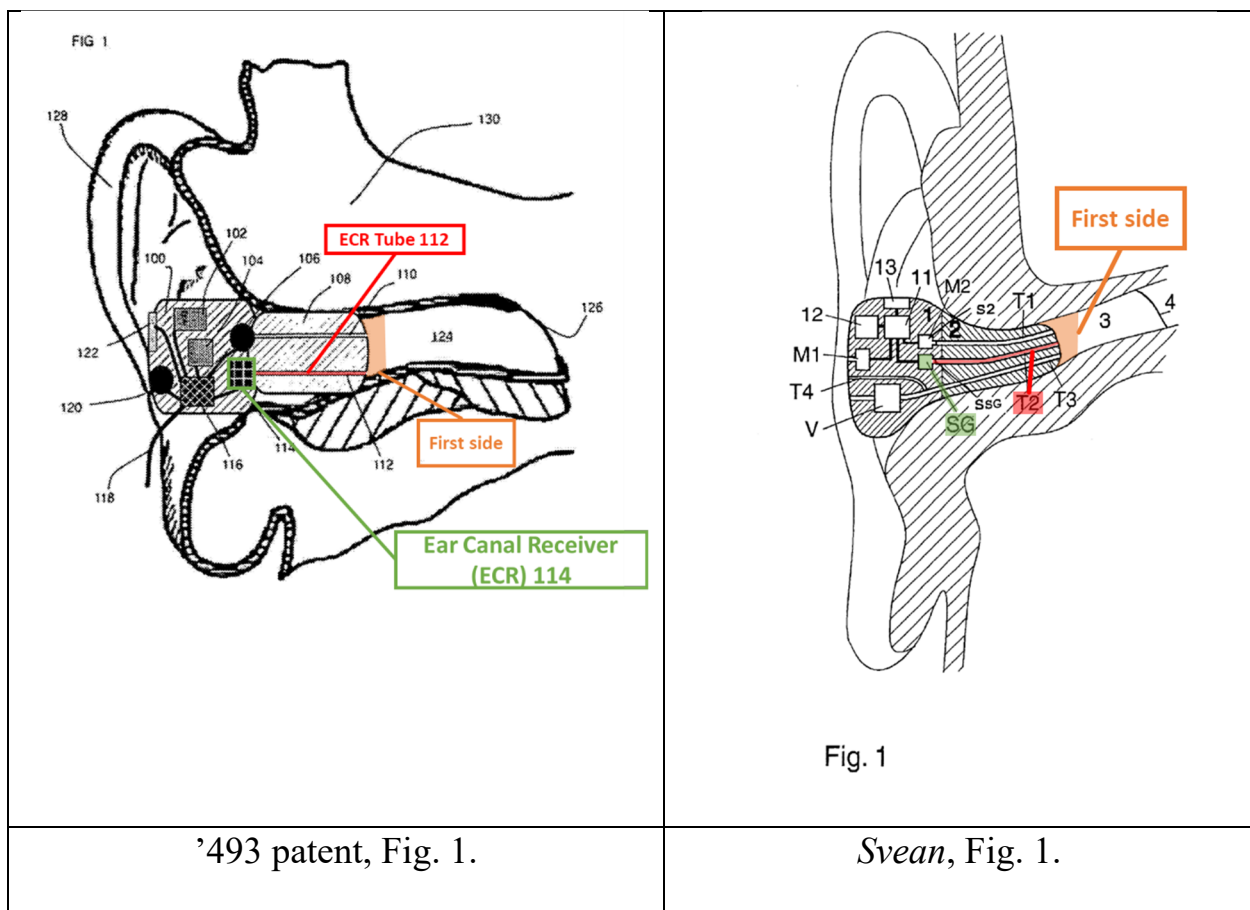
Svean, 11:65-12:6.

209. Similarly, the '493 patent discloses:

ECR 114 emits the test signal into Ear Canal Receiver Tube (**ECR Tube 112**). The test signal propagates through **ECR tube 112** and into ear canal 124.

'493 patent, 8:51-54.

210. In my opinion, the below comparison of the '493 patent and *Svean* demonstrates that a person having ordinary skill in the art would have understood that *Svean* discloses the speaker **SG** playing the test signal from the **first side** in the same way as the '493 patent.



211. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to send the test signal to **speaker SG** that plays the test signal from the side adjacent to the meatus of the ear canal (“the speaker plays the test signal from the first side”).

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

212. In my opinion, the *Svean-Voix-Padhi* combination discloses or suggests “[t]he audio device according to claim 4, wherein the first microphone measures sound from the first side.”

213. This limitation requires that the first microphone is located on the **first side** (i.e., inside the ear canal) and measures sound from this location..

214. *Svean* discloses that it is “possible to mount the **microphone M2** ... at the innermost part of the sealing section.” *Id.*, 6:22-26.

When smaller **microphones M2**, and **sound generators SG** are available, *it will be possible to mount the microphone M2 and the sound generator SG at the innermost part of the sealing section.*

Then there is no need for the transmission channels T1 and T2.

Svean, 6:22-26.

215. In that case, it is my opinion that **microphone M2** is a “first microphone” located on the **first side** of the earphone and “measures sound from” that location.

216. In addition, *Svean* discloses in Fig. 1 that **microphone M2** is connected through a **transmission channel T1** to the inner portion of the meatus 3 so that **microphone M2** measures sound that propagates from the **first side**. *Svean*, 12:33-36; *id.*, 11:65-12:6; *id.*, 12:58-61.

An acoustic **transmission channel T1** connects **microphone M2** to the inner portion of the meatus 3. **Microphone M2** picks up the sound field being set up by the **sound generator SG**.

Svean, 12:33-36.

A small **electro-acoustic transducer (sound source) SG** and a **microphone M2** are arranged in a **sealing section 2** arranged for attenuating sounds entering the meatus cavity 3. A **digital signal processing (DSP) or microcomputer unit E3** in the main section 1 or in the **sealing section 2** is used to generate a predetermined signal which is D/A converted by the D/A converter E7, amplified by amplifier E6 and applied to the **sound source SG**, *which generates a sound field in the closed part of the meatus 3*.

Svean, 11:65-12:6.

The resulting sound field is picked up by the microphone M2, amplified, converted to digital form and analysed by algorithms in the

processing unit E3 for a series of detectors represented by blocks 84, 85, 86 and 87.

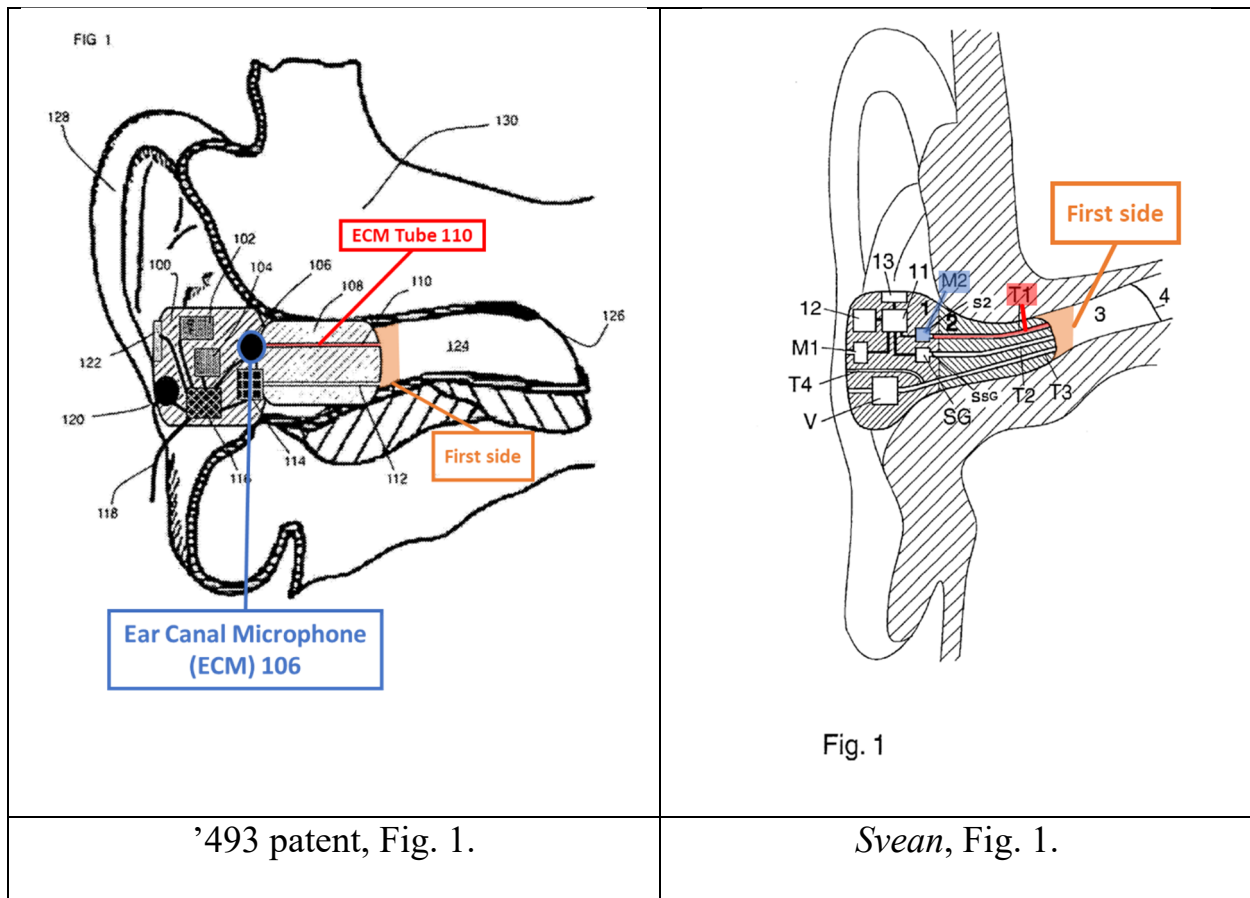
Svean, 12:58-61.

217. Similarly, the '493 patent discloses:

Ear Canal Microphone tube 110 (**ECM tube 110**) is configured to receive an acoustic signal incident on the **first side** of sealing unit 108. The test signal in ear canal 124 propagates through **ECM tube 110** and is received by **Ear Canal Microphone 106 (ECM 106)**. **ECM 106** is configured to *measure the test signal in ear canal 124* and provide the measured test signal to **processor 116**.

'493 patent, 8:54-60.

218. In my opinion, the below comparison of the '493 patent and *Svean* demonstrates that a person having ordinary skill in the art would have understood that *Svean* discloses that **microphone M2** measures sound from the **first side** in the same way as the '493 patent.



219. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to send a test signal to **speaker SG** that plays the test signal from the side adjacent to the meatus of the ear canal (“speaker plays the test signal from the first side” (claim 4)) and to receive a microphone signal from **microphone M2** that measures sound from the side adjacent to the meatus of the ear canal (“first microphone measures sound from the first side”).

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

220. In my opinion, the *Svean-Voix-Padhi* combination discloses or suggests “[t]he audio device according to claim 4, wherein the first microphone measures sound from the second side.”

221. This limitation requires that the first microphone is located on the **second side** (i.e., the side adjacent to the ambient environment) and measures sound from this location.

222. *Svean* discloses with respect to Fig. 1 that **microphone M1** is physically located on the **second side** (i.e., the side adjacent the ambient environment) and **processing unit E3** may receive a signal from **microphone M1** that measures “external”/“ambient” sound from the **second side**. *Svean*, 5:66-6:1; *id.*, 6:10-12; *id.*, 7:6-10; *see also* §X.A.1.c(iii).

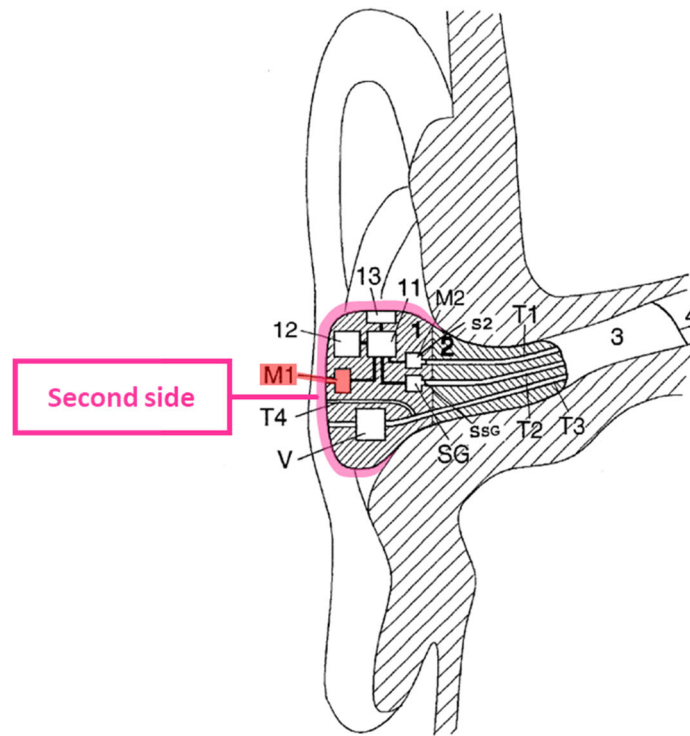


Fig. 1

Svean, Fig. 1.

FIG. 1 illustrates an embodiment according to the invention. The earplug comprises a main section 1 containing two microphones **M1** and **M2** and a **sound generator SG**.

Svean, 5:66-6:1.

The sound inlet of **microphone M1** is connected to the outside of the earplug, picking up the external sounds.

Svean, 6:10-12.

The **microphone M1** picks up the ambient sound. A signal from the **microphone M1** is amplified in E1 and sampled and digitised in an analogue to digital converter E2 and fed to a **processing unit E3** that

may be a digital signal processor (DSP), a microprocessor (uP) or a combination of both.

Svean, 7:6-10.

223. Regarding limitation [1.2.4] (“calculating a value using a first signal and a second signal...”), the same analysis I described in §X.A.1.c(iv) applies—that is, where the **microphone M1 signal** is the “first microphone signal,” the **microphone M2 signal** and the test signal are still the claimed “first signal” and “second signal” used in the calculation. I note that claims 1, 4, and 6 do not require that the “first microphone signal” received in [1.2.3] is used in the calculation.

224. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to send a test signal to **speaker SG** that plays the test signal from the side adjacent to the meatus of the ear canal (“speaker plays the test signal from the first side” (claim 4)) and to receive a signal from the **microphone M1** that measures sound from the side adjacent to the ambient environment (“first microphone measures sound from the second side”).

6. Claims 7, 8, 10, 11

225. These claims require that the first/second signal are “the” test signal/first microphone signal or a “modified” version of them. In the *Svean-Voix-*

Padhi combination, these signals are “modified” if they are pre-processed in *Padhi*’s **section 110** before performing cross correlation in **block 134 of section 112**.

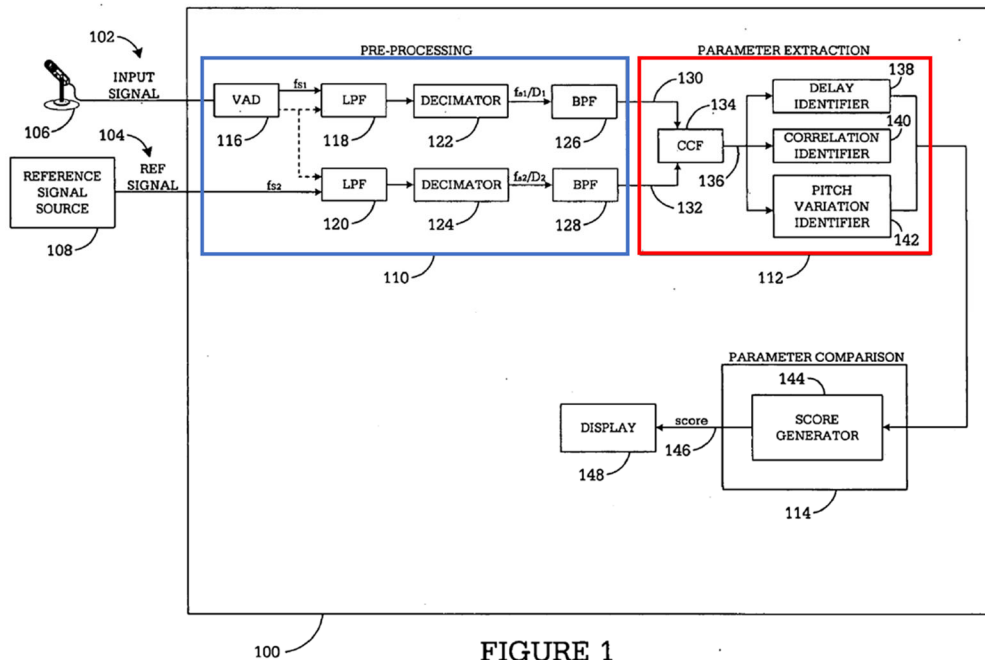


FIGURE 1

Padhi, Fig. 1.

226. As to each claim, it is my opinion that a person having ordinary skill in the art would have found that using “the” signals or a “modified” version of them amounts to a simple substitution of known elements according to known techniques to yield predictable results. 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.

227. 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 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**.

228. Further, 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 with 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].

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].

229. Moreover, it is my opinion that a person having ordinary skill in the art would have understood that the advantages and tradeoffs of making the substitutions were predictable, and the decision whether to implement one option or another would simply depend on the context. For example, omitting pre-processing would have the advantage of being more efficient/less computationally intensive.

230. 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**, such that the signal is “modified.” 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.

231. 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**.

a. 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.”

232. In my opinion, the *Svean-Voix-Padhi* combination discloses or suggests “[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.”

233. I first note that, while *Padhi* discloses that its [section 110](#) pre-processes audio signals prior to supplying them to its **cross correlation function 134**, in my opinion, this pre-processing is consistent with the requirement that either or both of the test and first microphone signals, themselves, be “used” in the cross-correlation calculation, as recited in claim 1, as a person or ordinary skill in the art would have understood that the cross-correlation is ultimately obtained by “using” the original test and microphone signals.

234. Regardless, even assuming the cross-correlation must be performed directly on the unmodified test and first microphone signals, in my opinion, the *Svean-Voix-Padhi* combination discloses or suggests supplying *Svean*'s pseudorandom sequence test signal and *Svean*'s [microphone M2 signal](#) to *Padhi*'s

cross correlation function 134 without first performing pre-processing on them in **section 110** to “modify” them.

235. In my opinion, it would have been obvious to use these signals in the *Svean-Voix-Padhi* combination for the reasons I explained in §X.A.6.

236. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to calculate a value that is the result of a non-phase cross correlation calculation (α) using “the” pseudorandom sequence test signal and “the” **microphone M2 signal** (“wherein the first signal is the test signal and the second signal is the first microphone signal”).

b. 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.”

237. In my opinion, the *Svean-Voix-Padhi* combination discloses or suggests “[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.”

238. As I discussed in §X.A.6.a, a person of ordinary skill in the art would have understood that supplying *Svean*’s pseudorandom test signal to *Padhi*’s **cross correlation function 134** after pre-processing the signal in **section 110** is

consistent with the requirement that the cross-correlation calculation “use” the unmodified test signal.

239. Regardless, even assuming cross-correlation must be performed directly on the unmodified test signal and on the modified first microphone signal, in my opinion, the *Svean-Voix-Padhi* combination discloses or suggests supplying *Svean*’s pseudorandom sequence test signal with at least one component in the claimed frequency range to *Padhi*’s **cross correlation function 134** without first performing pre-processing on it in *Padhi*’s **section 110** to “modify” it.

240. In my opinion, the *Svean-Voix-Padhi* combination also discloses or suggests supplying *Svean*’s **microphone M2 signal** to *Padhi*’s pre-processing **section 110**—creating a “modified” version of it with, among other operations, a low pass filter—and supplying a “modified” version of the **microphone M2 signal** to *Padhi*’s **cross correlation function 134**.

241. In my opinion, it would have been obvious to use these signals in the *Svean-Voix-Padhi* combination for the reasons I explained in §X.A.6.

242. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to calculate a value that is the result of a non-phase cross correlation calculation (α) using “the” pseudorandom sequence test signal and “a

modified” **microphone M2 signal** (“wherein the first signal is the test signal and the second signal is a modified first microphone signal”).

c. 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.”

243. In my opinion, the *Svean-Voix-Padhi* combination discloses or suggests “[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.”

244. As I discussed in §X.A.6.a, a person of ordinary skill in the art would have understood that supplying *Svean*’s first microphone signal to *Padhi*’s **cross correlation function 134** after pre-processing the signal in **section 110** is consistent with the requirement that the cross-correlation calculation “use” the unmodified first microphone signal

245. Regardless, even assuming that the cross-correlation must be performed directly on the first microphone signal, in my opinion, the *Svean-Voix-Padhi* combination discloses or suggests supplying *Svean*’s pseudorandom sequence test signal with at least one frequency component in the claimed frequency range to *Padhi*’s **pre-processing section 110**—creating a “modified” version of it with, among other operations, a low pass filter—and supplying a “modified” pseudorandom sequence test signal to *Padhi*’s **cross correlation function 134**.

246. Further, in my opinion, the *Svean-Voix-Padhi* combination also discloses or suggests supplying *Svean*'s **microphone M2 signal** to *Padhi*'s **cross correlation function 134** without first performing pre-processing on it in *Padhi*'s **section 110** to “modify” it. For example, if the “first microphone signal” is a narrow band audio signal, a person of ordinary skill in the art may wish to filter the full-spectrum pseudorandom sequence to match the frequency range of the audio, which would reduce processing needs.

247. In my opinion, it would have been obvious to use these signals in the *Svean-Voix-Padhi* combination for the reasons I explained in §X.A.6.

248. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to calculate a value that is the result of a non-phase cross correlation calculation (α) using a “modified” pseudorandom sequence test signal and “the” **microphone M2 signal** (“wherein the first signal is a modified test signal and the second signal is the first microphone signal”).

d. 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.”

249. In my opinion, the *Svean-Voix-Padhi* combination discloses or suggests “[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.”

250. In my opinion, the *Svean-Voix-Padhi* combination discloses or suggests supplying *Svean*’s pseudorandom sequence test signal with at least one frequency component in the claimed frequency range and *Svean*’s **microphone M2 signal** 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**.

251. In my opinion, it would have been obvious to use these signals in the *Svean-Voix-Padhi* combination for the reasons I explained in §X.A.6.

252. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to calculate a value that is the result of a non-phase cross correlation calculation (α) using a “modified” pseudorandom sequence test signal and a “modified” **microphone M2 signal** (“wherein the first signal is a modified test signal and the second signal is a modified first microphone signal”).

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

253. In my opinion, the *Svean-Voix-Padhi* combination discloses or suggests “[t]he audio device according to claim 1, wherein the earphone includes a second microphone.”

254. *Svean* discloses an ear terminal having at least two microphones. Claim 1 recites a “first microphone,” and in my opinion either of *Svean*’s **microphone M1** or **microphone M2** can satisfy the “first microphone” limitation. *See* §X.A.1.c(ii).

255. *Svean*’s **microphone M1** and **microphone M2** can be seen in *Svean*’s Fig. 1:

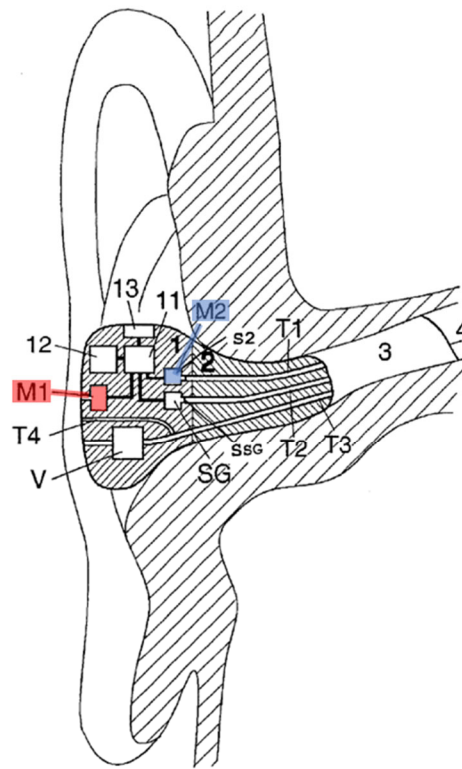


Fig. 1

Svean, Fig. 1.

256. Thus, in my opinion, if *Svean*'s **microphone M2** is the claimed "first microphone," then *Svean*'s **microphone M1** corresponds to the "second microphone" recited in claim 13. Conversely, if *Svean*'s **microphone M1** is the claimed "first microphone," then *Svean*'s **microphone M2** corresponds to the "second microphone" recited in claim 13.

257. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest

an ear terminal that includes a “second microphone” (“wherein the earphone includes a second microphone”).

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

258. In my opinion, the *Svean-Voix-Padhi* combination 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.”

259. When *Svean*’s **microphone M1** corresponds to the claimed “second microphone,” *Svean* discloses that the **processing unit E3** may receive a second microphone signal from **second microphone M1** corresponding to ambient/external sounds. *Svean*, 7:6-10; *id.*, 5:66-6:1; *id.*, 6:10-12; *id.*, Fig. 2.

The **microphone M1** picks up the ambient sound. A signal from the **microphone M1** is amplified in E1 and sampled and digitised in an analogue to digital converter E2 and *fed to a processing unit E3* that may be a digital signal processor (DSP), a microprocessor (uP) or a combination of both.

Svean, 7:6-10.

FIG. 1 illustrates an embodiment according to the invention. The earplug comprises a main section 1 containing two microphones **M1** and **M2** and a **sound generator SG**.

Svean, 5:66-6:1.

The sound inlet of **microphone M1** is connected to the outside of the earplug, picking up the external sounds.

Svean, 6:10-12.

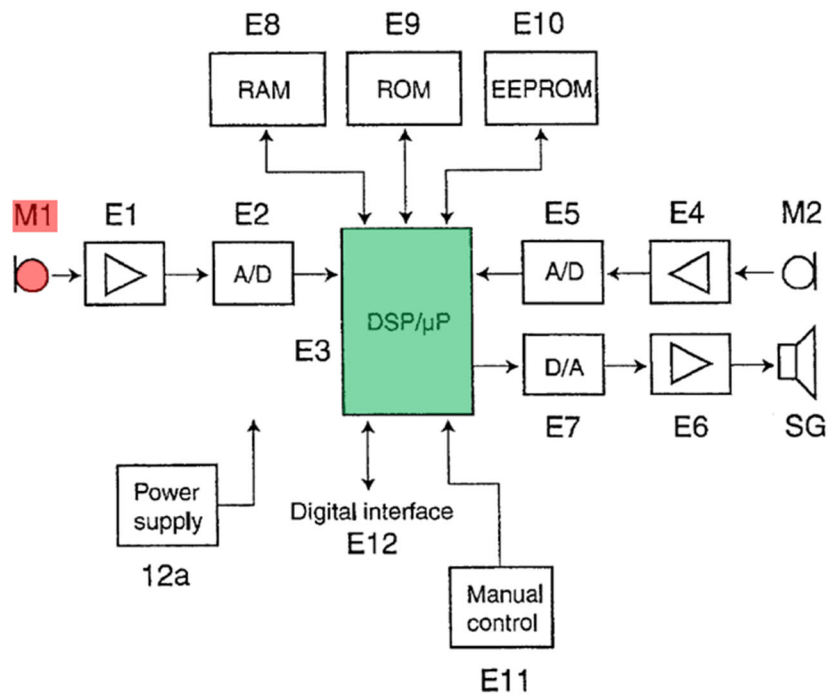


Fig. 2

Svean, Fig. 2.

260. As I explained in §X.A.1.c(iii), a person having ordinary skill in the art would have understood and found it obvious that *Svean* would use **microphone M1** to collect external sounds—and **processing unit E3** would be configured to receive the collected sounds—while performing the “verification” embodiment.

261. Conversely, when *Svean*'s **microphone M2** corresponds to the claimed "second microphone," *Svean* discloses that the **processing unit E3** may receive a second microphone signal from **second microphone M2** corresponding to sound within the wearer's ear canal. *Svean*, 12:1-9; *id.*, 5:66-6:1; *id.*, 6:12-14; *id.*, 7:10-15; *id.*, Fig. 2.

A **digital signal processing (DSP) or microcomputer unit E3** in the main section 1 or in the **sealing section 2** *is used to generate a predetermined signal* which is D/A converted by the D/A converter E7, amplified by amplifier E6 and *applied to the sound source SG, which generates a sound field in the closed part of the meatus 3. The microphone M2 picks up the sound in the meatus cavity 3.* This signal is amplified by amplifier E4, A/D converted by A/D converter E5, and *analysed in the digital signal processor or microprocessor E3.*

Id., 12:1-9.

FIG. 1 illustrates an embodiment according to the invention. The earplug comprises a main section 1 containing two microphones **M1** and **M2** and a **sound generator SG**.

Svean, 5:66-6:1.

The **microphone M2** is connected to the inner portion of the meatus 3 by means of an acoustic transmission channel T1.

Svean, 6:12-14.

A signal 51 from **microphone M2**, which picks up the sound in the meatus 3 between the isolating section 2 and the tympanum 4, is amplified in the amplifier E4 and sampled and digitised in the analogue to digital converter E5 and fed to the **processing unit E3**.

Svean, 7:10-15.

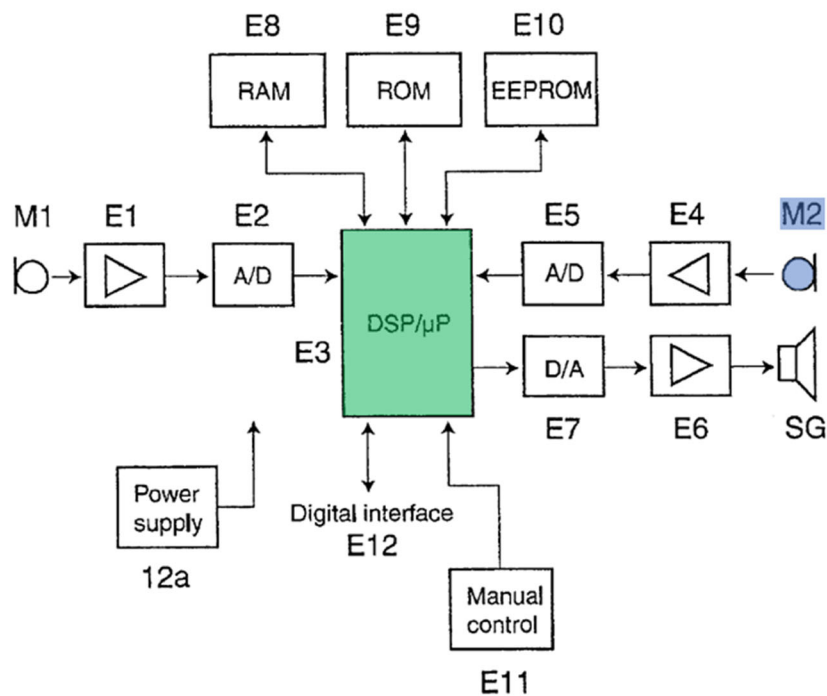


Fig. 2

Svean, Fig. 2.

262. In addition, I have been asked to confirm whether **processing unit E3's** microphone signal from **second microphone M2**, as I described above, reads on the '493 patent's disclosure of a processor that receiving a second microphone signal from the second microphone, and I confirm that it does. More particularly

the '493 patent discloses, with respect to steps 306 and 308 of the algorithm depicted in Figure 3, that, “[t]he 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.” ’493 patent, 8:9-13. Further, “ECM 106 is configured to measure the test signal in ear canal 124 and provide the measured test signal to processor 116.” *Id.*, 8:58-60.

263. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to receive a second microphone signal from the “second microphone” (“receiving a second microphone signal from the second microphone”).

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

264. In my opinion, the *Svean-Voix-Padhi* combination 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.”

265. As I explained previously, the seal quality is good when, for example, α is above a certain threshold. *See* §§X.A.1.c(v)-X.A.1.c(vi) (discussing limitations [1.2.5], [1.2.6], [1.2.7]).

266. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to compare a calculated value (α) to a threshold and to determine the seal quality is good when the value is at or above the threshold (i.e., either as binary result or a score) (“the seal quality is good if the value is at or above the threshold value”).

10. 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.”

267. In my opinion, the *Svean-Voix-Padhi* combination 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.”

268. In *Svean*, the **sealing section 2** may be made of slowly re-expanding shape polymer foam—i.e., a flexible polymer. *Svean*, 6:55-59; *id.*, 6:65-7:3.

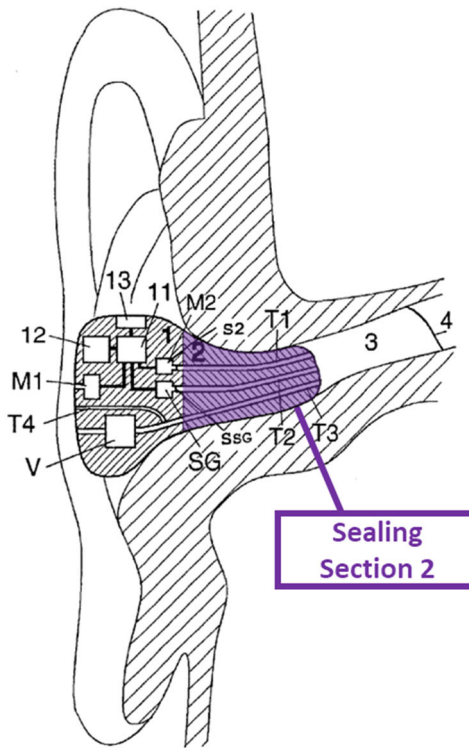


Fig. 1

Svean, Fig. 1.

The main section of the earplug may be made of standard polymer materials that are used for ordinary hearing aids. The **sealing part** *may be made of a resilient, slowly re-expanding shape retaining polymer foam like PVC, PUR or other materials suitable for earplugs.*

Svean, 6:55-59.

It is also possible to make the earplug in one piece comprising the main section 1 and the **sealing section 2**, *all made of a polymer foam mentioned above*, but then the channels T1, T2, T3, T4 have to be made of a wall material preventing the channels T1, T2, T3, T4 to collapse when the **sealing section 2** is inserted in the meatus 3.

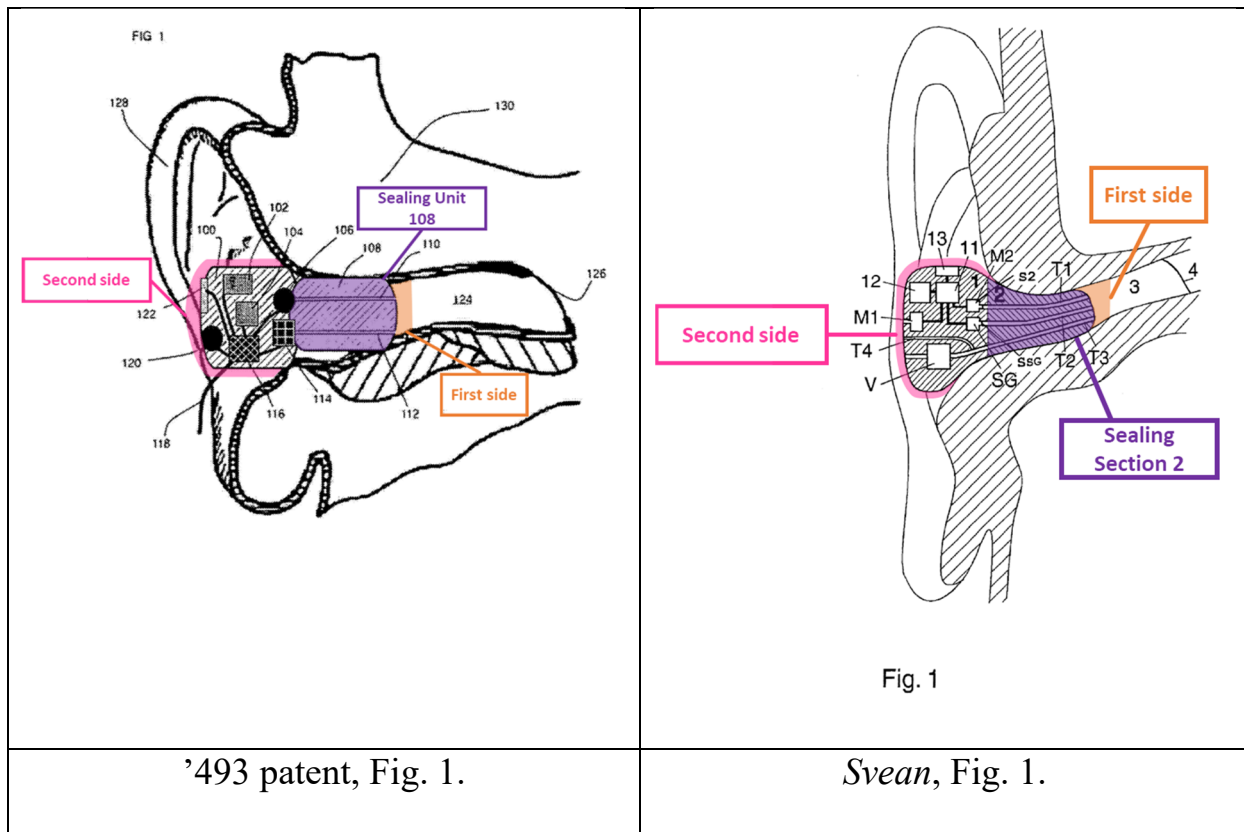
Svean, 6:65-7:3.

269. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that the **sealing section 2** of the ear terminal is composed of a flexible polymer (“wherein the sealing section is composed of at least one of rubber, silicone, a flexible polymer, or a combination thereof”).

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

270. In my opinion, the *Svean-Voix-Padhi* combination discloses or suggests “[t]he audio device of claim 1, wherein the sealing section is an eartip.”

271. The ’493 patent’s specification does not use the term “eartip.” In my opinion, as shown in the below comparison of Figs. 1 of the ’493 patent and *Svean*, *Svean* discloses an “eartip” in the same way the ’493 patent discloses an eartip—that is, a component that seals the **first side** of the ear terminal from the **second side**. Thus, in my opinion, a person having ordinary skill in the art would have understood that *Svean* discloses that the sealing section is an eartip in the same way as the ’493 patent.



272. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi* combination to disclose or suggest that **sealing section 2** of the ear terminal is an eartip (“sealing section is an eartip”).

B. Ground 2: *Svean*, *Voix*, *Padhi*, and *Hayashi* Render Obvious Claims 3 and 15-17

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

273. In my opinion, the *Svean-Voix-Padhi-Hayashi* combination 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.”

274. *Svean* discloses that “the user may get audible or other messaged confirmation if the leakage is acceptably low, or a warning signal if leakage is unacceptably high.” *Svean*, 12:11-14.

The user may get audible or other messaged confirmation if the leakage is acceptably low, or a warning signal if leakage is unacceptably high.

Svean, 12:11-14.

275. *Padhi* discloses generating a score that “is presented to a user or other person or people on a **display 148**.” *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].

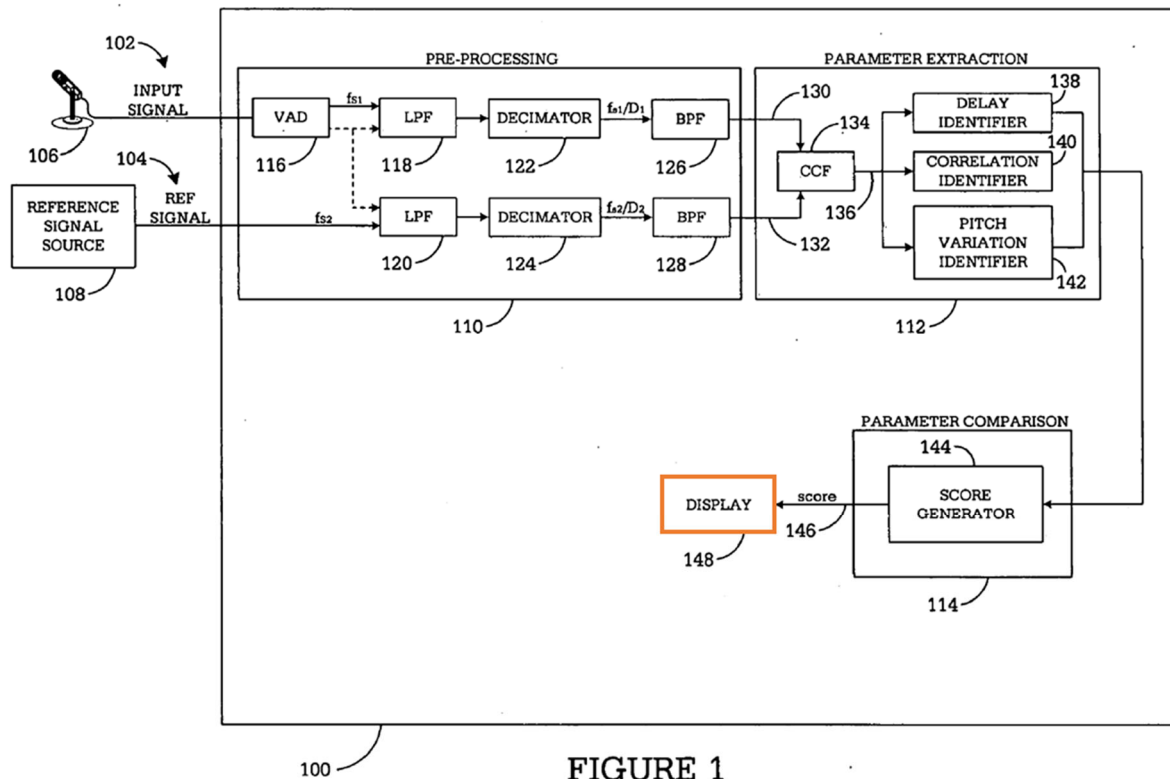
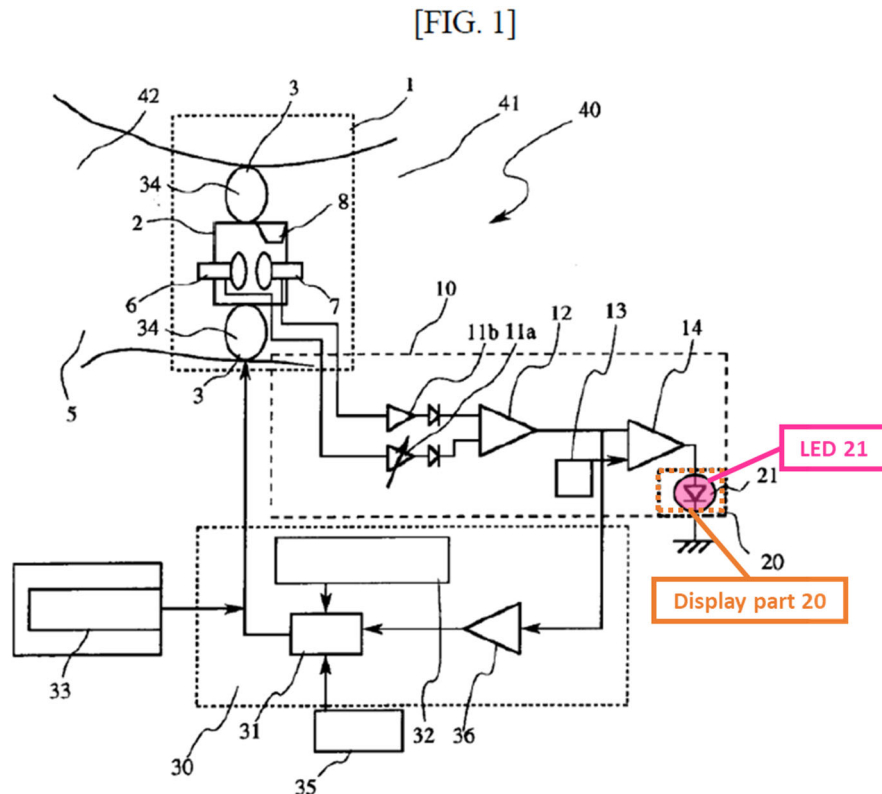


FIGURE 1

Padhi, Fig. 1.

276. Hayashi discloses determining the similarity between two microphone signals to evaluate the seal of an in-ear device. Hayashi's technique includes an LED 21 in display part 20 that indicates the level of sound leakage. Id., [0013]-[0014]; id., [0018]; id., [0020] (disclosing that "[a]ccording to this invention, a device having a function of objectively measuring and displaying a leakage amount of an external sound can be provided" and that "an external sound processing device such as a hearing aid having a function for displaying the leakage amount of sound ... can be acquired").



Hayashi, Fig. 1.

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].

[Effect of the Invention] *According to this invention, a device having a function of objectively measuring and displaying a leakage amount of an external sound can be provided. Furthermore, an external sound processing device such as a hearing aid having a function for displaying the leakage amount of sound, and having a small size suitable for a shape of an ear of a user and having a high soundproofing capability can be acquired.*

Hayashi, [0020].

277. In my opinion, a person having ordinary skill in the art would have been motivated to generate and send a visual message to a display of the audio device, in addition to or instead of sending an acoustic message conveying the seal quality.

278. First, to the extent the Board finds that *Svean*'s ear terminal may be the claimed "audio device" and seal quality may be a binary result, it is my opinion that a person having ordinary skill in the art would have been motivated to include *Hayashi*'s LED on *Svean*'s ear terminal to indicate a "good" or "bad" seal. *Svean* already contemplates providing "other messaged confirmation" regarding the seal quality (*Svean*, 12:11-14), and *Svean*'s "verification" algorithm is designed to be used in the immediate vicinity of a noisy environment (*id.*, 12:14-23) where it may be difficult to hear an acoustic message if the seal quality is poor.

The user may get audible or ***other messaged confirmation*** if the leakage is acceptably low, or a warning signal if leakage is unacceptably high. In the same manner, a signal may be transmitted to other instances, e.g. an external industrial health monitoring unit, with information about the leakage. ***One example may be that an ear terminal according to the invention is used for checking for leakage in the hearing protection while the wearer is at a gate controlling admittance to a noise exposed area.*** If leakage occurs, a signal may be transmitted from the ear terminal to a corresponding signal receiver at the gate, having means to block the gate for entrance until the leakage condition is remedied and verified.

Svean, 12:11-23.

279. In my opinion, a person having ordinary skill in the art would have been motivated to provide *Hayashi*'s LED display on the ear terminal to allow the user to view the result (e.g., in a mirror or other reflective surface), which—in

addition to a gate controlling access to the noisy environment (*Svean*, 12:19-23)—would serve as an additional measure to prevent the user from being exposed to noise with an inadequate seal.

280. Moreover, in my opinion, this would have the benefit of allowing another individual in the vicinity to view the LED and verify that the seal was adequate.

281. In addition, in view of the teachings of *Padhi* and *Hayashi*, depending on design needs, in my opinion a person having ordinary skill in the art would also have been motivated to provide a display such as a two-digit LED display on the ear terminal that provides a more granular score that the user (and optionally, another individual) could view to assess the seal before the user enters the environment.

282. Second, to the extent the Board finds that the “audio device” must be a separate structure, it is my opinion that a person having ordinary skill in the art would have been motivated to generate and send a visual message to the separate audio device in *Svean* to indicate either a binary result or, depending on design needs, a score in view of *Hayashi* and *Padhi*. For the same reasons I discussed above, it is my opinion that a person having ordinary skill in the art would have recognized that displaying the binary result or score on a separate audio device

would allow the user to view the result/score and ensure that the user received the message.

283. In either case, it is my opinion that a person having ordinary skill in the art would have had a reasonable expectation of success in implementing well-known display technology as described in *Hayashi* and *Padhi* to display the seal quality.

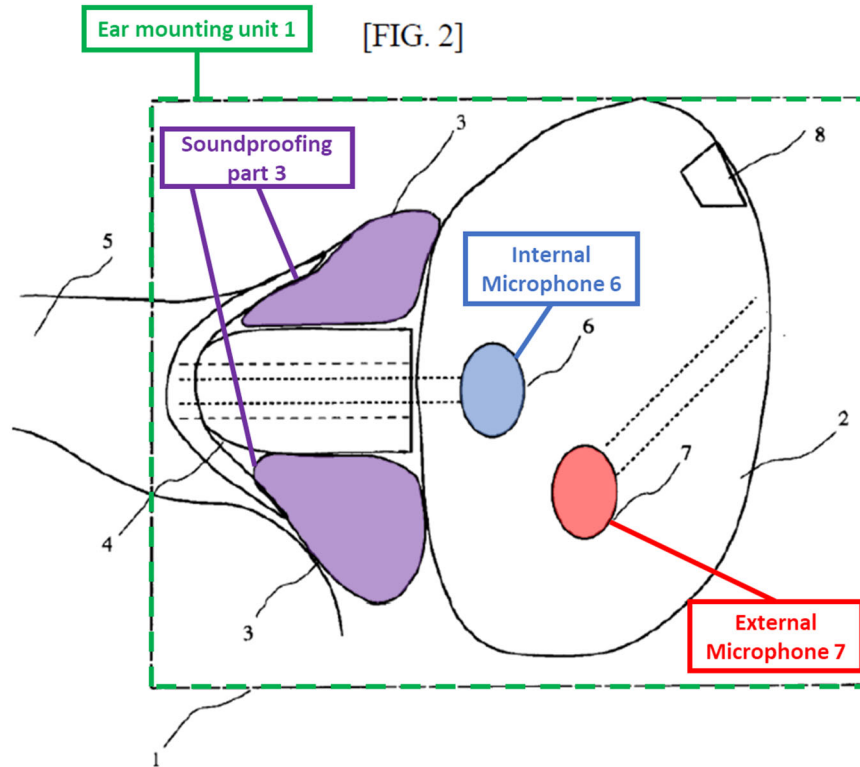
284. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi-Hayashi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to generate and send a visual message to a display of the ear terminal (or separate structure) to convey the seal quality—i.e., the binary result and/or *Padhi*'s score (“the message is a visual message sent to a display of the audio device”).

2. Claim 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.”

285. In my opinion, the *Svean-Voix-Padhi-Hayashi* combination discloses or suggests “[t]he audio device according to claim 14, wherein the first signal is the first microphone signal and the second signal is the second microphone signal.”

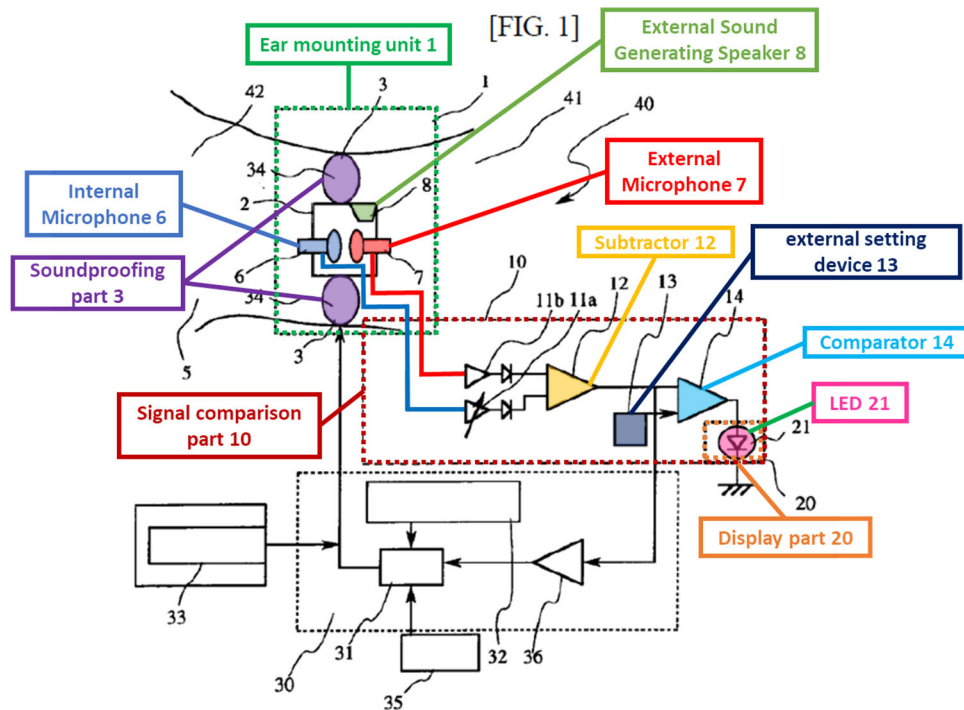
286. *Hayashi* discloses determining the similarity between two microphone signals to evaluate the seal of an in-ear device. In Figure 2, *Hayashi* discloses an

ear mounting unit 1 with an external microphone 7, internal microphone 6, and a soundproofing part 3.



Hayashi, Fig. 2.

287. *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.

288. 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].

289. *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].

290. 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.

291. In my opinion for the *Svean-Voix-Padhi-Hayashi* combination for limitation [1.2.4], *Svean* 's **microphone M2 signal** and **microphone M1 signal** would be inputs to *Padhi* 's **cross correlation function 134** either (1) in addition to the calculation performed with *Svean* 's pseudorandom test signal and the **microphone M2 signal**; or (2) instead of the pseudorandom test signal and the **microphone M2 signal**.

292. 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.

293. As to (1), in view of *Hayashi*, in my opinion a person having ordinary skill in the art would have been motivated to use two microphone signals in an additional calculation to determine seal quality because a person having ordinary skill in the art would have recognized that performing such a calculation would

provide further assurance that the seal quality was adequate. In my opinion, a person having ordinary skill in the art would have recognized that performing this additional calculation would be important for *Svean*'s ear terminal, because an erroneous seal quality determination may result in the user being exposed to unsafe noise levels with an inadequate seal.

294. Alternatively, as to (2), in my opinion a person having ordinary skill in the art would have been motivated to perform the calculation with *Svean*'s **microphone M1 signal** and **microphone M2 signal** instead of performing the calculation with the test signal and the **microphone M2 signal**. In my opinion, a person having ordinary skill in the art would have recognized that performing the calculation with two microphone signals was one of three predictable solutions that would work equally well for determining seal quality. The other two predictable solutions would be performing the calculation with: (i) the **microphone M2 signal** and the test signal; and (ii) the **microphone M1 signal** and the test signal. In my opinion, a person having ordinary skill in the art would have been motivated to choose one of these three predictable solutions to achieve the claimed invention.

295. In my opinion, the *Svean-Voix-Padhi-Hayashi* combination also discloses or suggests supplying *Svean*'s microphone signals output by *Svean*'s **microphone M2** and **microphone M1** to *Padhi*'s **cross correlation function 134** without first performing pre-processing on them in **section 110** to “modify” them.

296. In my opinion, it would have been obvious to use “unmodified” microphone signals in the calculation performed in the *Svean-Voix-Padhi-Hayashi* combination for the same reasons I explained in §X.A.6.

297. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi-Hayashi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to calculate a value that is the result of a non-phase cross correlation calculation (α) using “the” **microphone M2 signal** and “the” **microphone M1 signal** (“wherein the first signal is the first microphone signal and the second signal is the second microphone signal”).

3. Claim 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.”

298. In my opinion, the *Svean-Voix-Padhi-Hayashi* combination discloses or suggests “[t]he audio device of claim 14, wherein the first signal is a modified first microphone signal and the second signal is the second microphone signal.”

299. For the reasons I explained with respect to claim 15, it would have been obvious in view of *Hayashi* to use *Svean*’s **microphone M1 signal** and **microphone M2 signal** in *Padhi*’s calculation. See §X.B.2.

300. In my opinion, the *Svean-Voix-Padhi-Hayashi* combination discloses or suggests supplying *Svean*’s **microphone M2 signal** to *Padhi*’s **pre-processing**

section 110—creating a “modified” version of it with, among other operations, a low pass filter—and supplying a “modified” **microphone M2 signal** to *Padhi*’s **cross correlation function 134**.

301. In my opinion, *Svean* and *Padhi* also disclose or suggest supplying *Svean*’s **microphone signal M1** to *Padhi*’s **cross correlation function 134** without first performing pre-processing on it in *Padhi*’s **section 110** to “modify” it.

302. In my opinion, it would have been obvious to use these signals in the *Svean-Voix-Padhi-Hayashi* combination for the reasons I explained in §X.A.6.

303. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi-Hayashi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to calculate a value that is the result of a non-phase cross correlation calculation (such as α) using a “modified” **microphone M2 signal** and “the” **microphone M1 signal** (“wherein the first signal is a modified first microphone signal and the second signal is the second microphone signal”).

4. Claim 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.”

304. In my opinion, the *Svean-Voix-Padhi-Hayashi* combination discloses or suggests “[t]he 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.”

305. For the reasons I explained with respect to claim 15, it would have been obvious in view of *Hayashi* to use *Svean*’s **microphone M2 signal** and **microphone M1 signal** in *Padhi*’s calculation. See §X.B.2.

306. In my opinion, the *Svean-Voix-Padhi-Hayashi* combination discloses or suggests supplying *Svean*’s **microphone M2 signal** to *Padhi*’s **pre-processing section 110**—creating a “modified” version of it with, among other operations, a low pass filter—and supplying a “modified” **microphone M2 signal** to *Padhi*’s **cross correlation function 134**.

307. In my opinion, *Svean* and *Padhi* also disclose or suggest supplying *Svean*’s **microphone M1 signal** to *Padhi*’s **pre-processing section 110**—creating a “modified” version of it with, among other components, a low pass filter—and supplying a “modified” **microphone M1 signal** to *Padhi*’s **cross correlation function 134**.

308. In my opinion, it would have been obvious to use these signals in the *Svean-Voix-Padhi-Hayashi* combination for the reasons I explained in §X.A.6.

309. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi-Hayashi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one

of the **memory elements** to calculate a value that is the result of a non-phase cross correlation calculation (such as α) using a “modified” **microphone M2 signal** and a “modified” **microphone M1 signal** (“the first signal is a modified first microphone signal and the second signal is a modified second microphone signal”).

C. Ground 3: *Svean, Voix, Padhi, Hayashi, and Bourk* Render Obvious Claims 9 and 12

- 1. 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.”**

310. In my opinion, the *Svean-Voix-Padhi-Bourk* combination 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.”

311. 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.

312. 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.

313. 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.

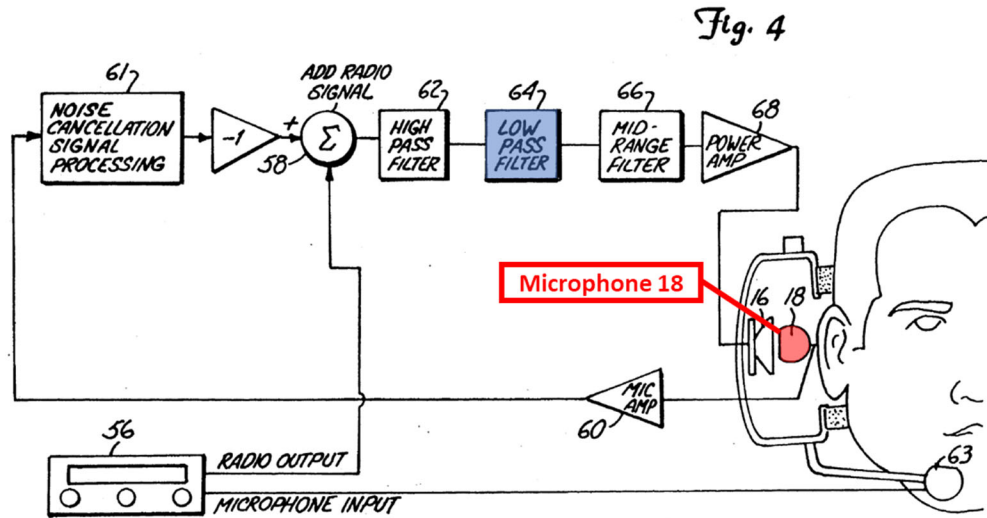
Bourk, 6:25-35.

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.

314. 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.

315. In my opinion, in view of *Bourk*, a person having ordinary skill in the art would have found applying a low pass filter to *Svean*'s **microphone M2 signal** 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*).

316. Thus, in the *Svean-Voix-Padhi-Bourk* combination, *Svean*'s **microphone M2 signal** 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**.

317. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi-Bourk* combination to disclose or

suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to calculate a value that is the result of a non-phase cross correlation calculation (α) using “the” pseudorandom sequence test signal and “a modified” **microphone M2 signal** (“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 the *Svean-Voix-Padhi-Bourk* combination to disclose or suggest that the “modified” **microphone M2 signal** 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”).

2. 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.”

318. In my opinion, the *Svean-Voix-Padhi-Bourk* combination 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.”

319. 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* §X.C.1. Further, in my opinion, a person having ordinary skill in the art would have found applying a low pass filter to *Svean*’s **microphone M2 signal** with an upper cutoff frequency less than or equal to 200 Hz amounts to nothing more than a simple substitution of known elements according to known techniques to yield predictable results. *Id.*

320. Accordingly, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Padhi-Voix-Padhi* combination to disclose or suggest that **processing unit E3** is configured to execute instructions stored in one of the **memory elements** to calculate a value that is the result of a non-phase cross correlation calculation (such as α) using a “modified” pseudorandom sequence test signal and a “modified” **microphone M2 signal** (“wherein the first signal is a modified test signal and the second signal is a modified first microphone signal”) as required by claim 11. Further, in my opinion, a person having ordinary skill in the art would have understood the *Svean-Voix-Padhi-Bourk* combination to disclose or suggest that the “modified” **microphone M2 signal** 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 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”).

XI. SECONDARY CONSIDERATIONS

321. 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.

XII. CONCLUSION

322. The findings and opinions set forth in this declaration are based on my work and examinations to date.

323. 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.

324. 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.

325. 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: June 21, 2024

A handwritten signature in black ink, appearing to read 'Chris Kyriakakis', written over a horizontal line.

Chris Kyriakakis, Ph.D.