UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

K/S HIMPP
Petitioner

v.

III Holdings 4 LLC
Patent Owner.

Case No. TBD

DECLARATION OF LES ATLAS, PH.D.
REGARDING U.S. PATENT NO. 8,654,999

HIMPP 1008

TABLE OF CONTENTS

Page

I, Les Atlas, Ph.D., declare as follows:

I. INTRODUCTION

1. My name is Les Atlas, Ph.D. I have been asked to opine on the patentability of U.S. Patent 8,654,999 by Mindlin et al. ("the '999 patent"), entitled "System and Method of Progressive Hearing Device Adjustment." My opinions are set forth herein. I make this declaration based on personal knowledge and I am competent to testify about the matters set forth herein. I submit this declaration in support of K/S HIMPP's Petition for *Inter Partes* Review, which I have read and fully support as if my own.

2. I have been retained by counsel for K/S HIMPP ("Petitioner") to serve as a technical expert in this *Inter Partes* Review proceeding. I have been asked to provide expert testimony in this declaration regarding the patentability of the claims of the '999 patent and the grounds of unpatentability upon which the *Inter Partes* Review petition are based.

3. I am being compensated for my time at my normal hourly rate of $270 and for reasonable expenses incurred in preparing this declaration.

4. My compensation is not dependent on and in no way affects the substance of my statements in this Declaration.

5. I have no financial interest in the Petitioner. I similarly have no financial interest in the '999 patent.

II. QUALIFICATIONS

6. My academic credentials include a B.S. in Electrical Engineering from the University of Wisconsin and a M.S. and a Ph.D. in Electrical Engineering from Stanford University. I am a Fellow of the Institute of Electrical and Electronics Engineers (IEEE) and have been and remain active as an electrical engineering, hearing, and speech science university faculty educator and researcher. My work and impact in hearing research goes back to 35 years ago when I designed the world's first portable speech processor for cochlear implants. This then-new technology was like a hearing aid, except that it used electrical stimulation of the inner ear to treat patients who were profoundly deaf, that is could not hear at all, even with a sound amplification hearing aid. This cochlear implant technology has since become a common form of treatment, and is used by over 190,000 users worldwide. Cochlear implant technology and regular hearing aids share challenges such as sound shaping, frequency filtering, and range of amplification, along with portability. That is, both have small external processors where sounds are conditioned, often with parameters which are customized for each patient. More recently, since about 2004, I have addressed the lack of rich music perception and challenges for speech perception with noisy background by both hearing aid and cochlear implant users. My innovations resulted in several key publications, such as the May 2008 issue of *Hearing Research*, where our

paper "Improving performance in noise for hearing aids and cochlear implants using coherent modulation filtering," was featured on the cover of this issue. The work described in this paper resulted in my 2012 Bloedel Scholar Award, given out by the Bloedel Speech and Hearing Research Institute. It also resulted in 2014-16 research grants from the Coulter Foundation. The approach described in the paper came from my decades of more theoretical work in time-frequency analysis. That work resulted in my election to the high level of Fellow of the IEEE "[f]or contributions to time-varying spectral analysis and acoustical signal processing." This approach was also used to modernize music coding for all listeners worldwide. Our coherent modulation approach resulted in my 2003 Fulbright Award, where I spent 6 months at the Fraunhofer Institute in Germany and then 3 months in Cambridge England. Since then (2003) my commitment to solving challenges facing perception of music and speech in noise in cochlear implant patients has resulted in more publications and progress in those needed research directions.

7. My latest curriculum vitae (CV) is attached to this declaration as an Appendix A.

III. BASIS OF OPINIONS

8. I have reviewed the specification and claims of the '999 patent. '999 patent, Ex. 1001.

9. I have also reviewed the following references, all of which I understand to be prior art to the ’999 patent:

- U.S. Patent No. 8,787,603 to Fichtl, et al. (“Fichtl,” Ex. 1003);
- U.S. Patent Application Publication 2003/0215105 to Sacha (“Sacha,” Ex. 1004);
- U.S. Patent Application Publication 2005/0036637 to Janssen (“Janssen,” Ex. 1005);
- U.S. Patent No. 6,741,712 to Bisgaard (“Bisgaard,” Ex. 1006);
- U.S. Patent No. 4,972,487 to Mangold (“Mangold,” Ex. 1007);
- German patent publication DE19542961, with certified translation (“DE ‘961,” Ex. 1009);
- U.S. Patent No. 7,933,419 to Roeck (“Roeck,” Ex. 1010).

10. In addition to the documents listed above, I have also reviewed parts of the file history of the ’999 patent, the accompanying petition, all of the documents listed in Petitioner’s List of Exhibits in the accompanying petition, and all of the documents cited in this Declaration.

IV. UNDERSTANDING OF LEGAL PRINCIPLES

11. I am not an attorney. For the purposes of this declaration, I have been informed about certain aspects of the law that are relevant to my opinions. My understanding of the law was provided to me by Petitioner’s attorneys.

12. I understand that prior art to the ’999 patent includes patents and printed publications in the relevant art that predate the priority date of the alleged

invention recited in the '999 patent. I have applied the date of April 13, 2010, the earliest possible filing date of the earliest provisional patent application to which the '999 patent claims priority, as the priority date, although the '999 patent may actually not be entitled to such an early priority date. My opinions regarding the '999 patent and the unpatentability of its claims are the same regardless of whether the earliest priority date of the '999 patent is April 13, 2010 (filing date of the provisional application no. 61/323,841), June 2, 2010 (filing date of provisional application no. 61/305,759), or April 12, 2011 (filing date of application no. 13/085,016).

13. I understand that a claim is unpatentable if it would have been obvious to a person of ordinary skill in the art ("POSA") at the time the alleged invention was made. I understand that a claim could have been obvious from a single prior art reference or from a combination of two or more prior art references.

14. I understand that an obviousness analysis requires an understanding of the scope and content of the prior art, any differences between the alleged invention and the prior art, and the level of ordinary skill in evaluating the pertinent art.

15. I further understand that certain factors may support or rebut the obviousness of a claim. I understand that such secondary considerations include, among other things, commercial success of the patented invention, skepticism of

those having ordinary skill in the art at the time of invention, unexpected results of the invention, any long-felt but unsolved need in the art that was satisfied by the alleged invention, the failure of others to make the alleged invention, praise of the alleged invention by those having ordinary skill in the art, and copying of the alleged invention by others in the field. I understand that there must be a nexus, that is, a connection, between any such secondary considerations and the alleged invention. I also understand that contemporaneous and independent invention by others is a secondary consideration tending to show obviousness.

16. I further understand that a claim would have been obvious if it unites old elements with no change to their respective functions, or alters prior art by mere substitution of one element for another known in the field, and that combination yields predictable results. Also, I understand that obviousness does not require physical combination/bodily incorporation, but rather consideration of what the combined teachings would have suggested to persons of ordinary skill in the art at the time of the alleged invention.

17. While it may be helpful to identify a reason for this combination, I understand that there is no rigid requirement of finding an express teaching, suggestion, or motivation to combine within the references. When a product is available, design incentives and other market forces can prompt variations of it, either in the same field or a different one. If a POSA can implement a predictable

variation, obviousness likely bars its patentability. For the same reason, if a technique has been used to improve one device and a POSA would recognize that it would improve similar devices in the same way, using the technique would have been obvious. I understand that a claim would have been obvious if a POSA would have had reason to combine multiple prior art references or add missing features to reproduce the alleged invention recited in the claims.

18. I am not aware of any allegations by the named inventors of the '999 patent or any assignee of the '999 patent that any secondary considerations tend to rebut the obviousness of any claim of the '999 patent discussed in this declaration.

19. I understand that in considering obviousness, it is important not to determine obviousness using the benefit of hindsight derived from the patent being considered.

20. I understand that other challenges to the validity of a patent, including patent ineligibility, enablement, written description, and definiteness, cannot be raised in IPR proceedings before the Board to challenge the validity of the '999 patent. Accordingly, I did not consider those other challenges.

21. The analysis in this declaration is in accordance with the above-stated legal principles.

V. DESCRIPTION OF THE RELEVANT FIELD AND RELEVANT TIMEFRAME

22. The ’999 patent was issued to Harold S. Mindlin, II, et al. on February 18, 2014. I have been informed that the ’999 patent claims priority to Provisional Application No. 61/323,841 filed on April 13, 2010 and Provisional Application No. 61/305,759 filed on June 2, 2010.

23. I have carefully reviewed the ’999 patent and portions of its file history.

24. Based on my review of this material, I believe that the relevant field for the purposes of the ’999 patent is hearing aid systems. I have been informed that the relevant time frame is before April 13, 2010, which is the filing date of U.S. Provisional Application No. 61/323,841, although the ’999 patent may actually not be entitled to April 13, 2010 as such an early priority date.

25. As described above and as shown in my CV, I have extensive experience in the relevant field. Based on my experience, I have a good understanding of the relevant field in the relevant timeframe and the skills possessed by those of ordinary skill at the time.

VI. THE PERSON OF ORDINARY SKILL IN THE RELEVANT FIELD IN THE RELEVANT TIMEFRAME

26. I understand that the level of ordinary skill may be reflected by the prior art of record, and that a POSA to which the claimed subject matter pertains

would have the capability of understanding the scientific and engineering principles applicable to the pertinent art. I understand that one of ordinary skill in the art has ordinary creativity, and is not a robot.

27. I understand there are multiple factors relevant to determining the level of ordinary skill in the art, including (1) the levels of education and experience of persons working in the field at the time of the invention, (2) the sophistication of the technology, (3) the types of problems encountered in the field; and (4) the prior art solutions to those problems. There are likely a wide range of educational backgrounds in the technology fields pertinent to the '999 patent.

28. The '999 patent relates to the technical field of hearing aid systems. More specifically, the field includes hearing aids that contain a processor, memory, and instructions for carrying out an acclimatization program as well as computer/software-based tools used by hearing care professionals, *e.g.*, audiologists, to fit hearing aids to users who have experienced partial or complete hearing loss. A POSA at the time of the alleged invention of the '999 patent would have had a B.S. degree in electrical or computer engineering, or the equivalent, and at least two years of experience in hearing aid systems. Graduate education could substitute for work experience, and additional work experience/training could substitute for formal education. As described in more detail above in ¶ 6, I would

have been a person with at least ordinary skill in the art of the ’999 patent as of the time of its alleged invention.

VII. TECHNOLOGY BACKGROUND FOR THE ’999 PATENT

29. The ’999 patent claims priority to Provisional Application No. 61/323,841 filed on April 13, 2010 and Provisional Application No. 61/305,759 filed on June 2, 2010, and I have been informed that the relevant time frame is before April 13, 2010, even though the ’999 patent may not be entitled to that

30. Before April 13, 2010, hearing aid technology was typically slightly less advanced when compared to today’s technology. The lowest cost hearing assistance devices would simply be a microphone wired to an analog amplifier which was then connected, by wire, to a small speaker in the ear canal. However, more sophisticated hearing aids which were readily available at the time of this April 13, 2010 relevant timeframe date would have had digital processing, via a processor or processors in the device, along with analog-to-digital and digital-to-analog converters, where the digital processing entailed extensive sound conditioning to reduce perceived noise to, ideally, make speech more intelligible.

31. These hearing aids would also allow extensive customized fitting by an audiologist to adjust parameters such as the amount of amplification as a function of frequency and the amount of amplification as a function of input level (*e.g.*, dynamic range compression), where one or multiple sets of signal processing

parameters were customized for each patient. This customization of hearing aid profiles for each patient can sometimes be called sound shaping, where this shaping refers to, for example, amplification as a function of frequency, amplification as a function of signal level (*e.g.*, dynamic range compression), and/or dynamics of the compression, such as attack and decay times. There are typically more than one set of parameter settings or hearing aid profiles for each patient, since these parameter settings might differ for listening to speech in quiet, speech in a noisy room, music listening, listening from broadcast material, such as television, and listening to speech over a telephone. Ex. 1012, http://www.isa-audiology.org/members/pdf/gpg-adaf.pdf (last visited on Jan. 3, 2016, published Nov. 2004). These parameter profiles were typically set initially from previous audiogram measurements by the audiologist on the patient in a special listening room with calibrated headphones, and subsequently adjusted during fitting of the hearing aid, with real-time sound or direct audio input provided to allow for this adjustment. Ex. 1011, http://www.audiologyonline.com/articles/guideline-for-audiologic-management-adult-966 (last visited on Jan. 4, 2016, published Oct. 30 2006); Ex. 1012.

32. Before April 13, 2010, it was recognized in the field of hearing aid systems that new hearing aid users had difficulty adjusting to a full, complete compensation for their hearing loss. '999 patent, Ex. 1001 at 1:57-67, 2:26-39;

Fichtl, Ex. 1003 at Abstract, 1:6-9, 1:19-2:18; Sacha, Ex. 1004 at [0003], [0012]. For new hearing aid users, a slow adjustment over time from uncompensated hearing to full, complete compensation is more comfortable and less jarring. *Id*.

33. In 2006, the American Academy of Audiology (AAA) developed a Guideline for Audiologic Management of the Adult Patient. Ex. 1011. The Guideline included a Hearing Aid Orientation to ensure patients obtained the desired benefits from amplification as easily and efficiently as possible. *Id*. at 5. The orientation information could be device or patient-related. *Id*. The patient-related issues included helping the patient adjust to amplification and to have realistic expectations of the benefits and limitations of amplification. *Id*. at 5-6. These issues can also relate to the use of acclimatization programs in order to help new patients adjust to wearing hearing aids. The Guideline recognized that successful management of the treatment process required comprehensive counseling to help patients adjust to their hearing aids. *Id*. at 6.

34. A paper was published in July 2009, Ex. 1013, which said that: "..hearing devices have been introduced that contain a feature that will automatically increase gain over a predetermined period of time (e.g. Robinson & Verberne, 2003; Schum & Beck, 2006)." This paper discusses hearing aid systems which automatically increase gain, that is, amplification. In terms of similar adaptation yet also for frequency filtering, a 2008 paper, Ex. 1014, page 32.

suggests that: "For example, the amount of amplification has to vary with frequency, and the amount at each frequency has to vary with input level, usually by different degrees. Therefore, the next level of sophistication for the trainable hearing aid is the individual adjustment of the shape of the gain-frequency response." This quote is referring to what a POSA would interpret as an adaptable amount of frequency filtering.

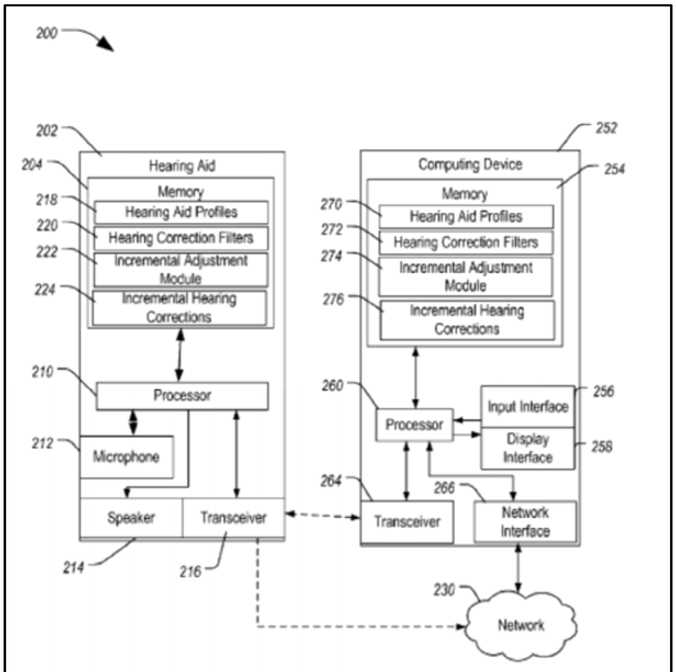
VIII. THE '999 PATENT (EX. 1001)

35. The '999 patent relates to a hearing aid system that includes the ability to provide an incremental or progressive hearing adjustment for a hearing aid user. '999 patent, Ex. 1001 at Abstract, 1:58-67, 2:26-39. This incremental or progressive hearing adjustment is frequently referred to as an acclimatization program.

36. A first-time hearing aid user may not be accustomed to hearing certain sounds or frequencies due to the hearing aid user's hearing loss. Some first-time hearing aid users may experience psychological distress when their hearing is restored to a normal level after years of hearing loss. '999 patent, Ex. 1001 at 1:58-67. A hearing aid user suffering from distress may stop wearing his or her hearing aid. *Id.* The hearing aid cannot do its job of providing compensation for a user's hearing loss unless the hearing aid user routinely wears and uses the hearing aid.

37. The ’999 patent relates to implementing incremental, progressive adjustments over time to accustom a hearing aid user from an uncompensated hearing level to a fully-compensated hearing level instead of an abrupt transition. ’999 patent, Ex. 1001 at 2:26-39. This type of progressive adjustment is referred to as an acclimatization program or a habituation program.

38. The ’999 patent discloses a hearing aid system that includes a hearing aid, which can include a transceiver, a processor, a microphone, a speaker, and a memory. ’999 patent, Ex. 1001 at 5:49-6:2; Fig. 2. The system may also include a computing device, which includes a transceiver, a processor, a display interface, an input interface, a network interface, and a memory. *Id*. at 6:13-41; Fig. 2. The hearing aid and computing device can communicate either through the transceivers or through a network. *Id*. at Fig. 2. The memory, in the hearing aid or in the computing device, can store instructions as well as hearing aid profiles, hearing correction filters, incremental hearing corrections, and incremental hearing correction filters. *Id*. at Fig. 2, 5:63-6:2, 6:29-41.

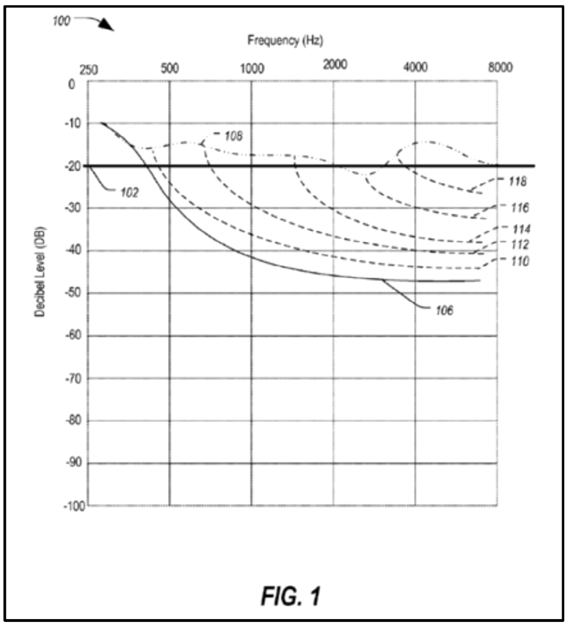


’999 patent, Ex. 1001, Figure 2

39. In operation, without any acclimatization adjustments, the ’999 patent’s hearing aid’s processor applies the user’s hearing aid profile, so shaped acoustic signals are provided to a speaker or bone conduction element. *Id*. at 2:44-46. Normally, the hearing aid processor would immediately apply the hearing aid profile. *Id*. at 4:48-59. The purported invention of the ’999 patent is the application of multiple, incremental correction levels in the form of hearing correction filters. *Id*. at 4:60-5:2, 6:42-52, 8:55-58. The filters are applied to the user’s hearing aid profile before producing an output signal. *Id*. Both the hearing corrections filters and the amount of time that a particular hearing correction filter

is applied before transitioning to the next hearing correction filter may vary. *Id.* at 8:36-54.

40. For example, Figure 1 is a graph of decibel level (in dB) versus frequency (in Hertz) that depicts a representative user's hearing deficit and a series of incremental adjustments to advance the user's hearing from the user's hearing deficit level to an adjusted hearing level associated with a hearing aid profile. '999 patent, Ex. 1001 at 3:56-63, Fig. 1.



'999 patent, Ex. 1001, Figure 1

The solid line 106 depicts the particular user's uncompensated hearing. *Id.* at 4: 2-3. The user's hearing aid profile is shown as line 108. *Id.* at Fig. 1, 4:15-22. In normal operation, the hearing aid will apply the user's hearing aid profile, which is customized to correct the user's hearing, to that the user's hearing sensitivity will correspond to the hearing aid profile correction line 108. *Id.* at 4:4-14. The hearing aid profile depicts varying decibel levels according to frequency, which are based not the particular user's hearing loss.

41. The lines 110, 112, 114, 116, and 118 in Figure 1 are intermediate hearing sensitivity levels that are achieved by applying hearing correction filters to the hearing aid profile or by applying incremental hearing corrections, or incremental hearing correction filters, to an audio signal. '999 patent, Ex. 1001 at 4:15-27. For example, the specification states, "[e]ach of the intermediate hearing sensitivity lines 110, 112, 114, 116, and 118 represents one or more adjustments to enhance the user's hearing sensitivity by applying an incremental hearing filter the selected hearing aid profile, reducing its hearing correction by a pre-determined amount." *Id.* at 4:22-27. These example lines show adjustments at selected frequencies to the desired hearing level while providing less of an enhancement to other frequencies. *Id*. at 4:35-47; *see also id.* at 5:39-48. However, other incremental hearing corrections can also be implemented, such as evenly adjusting the hearing correction across all frequencies or adjusting selected frequencies by different amounts. *Id.*

42. The hearing aid system disclosed in the 999 patent includes instructions that are executable by a processor of a computing device, a hearing aid, or another system in order to produce multiple correction levels, which can be applied to the selected hearing aid profile. '999 patent, Ex. 1001 at 4:60-5:2. The

multiple correction levels gradually adjust the selected hearing aid profile to allow a hearing aid user to become accustomed to the hearing aid and its audio compensation. *Id.*

43. In one embodiment of the hearing aid system, the hearing aid may generate a trigger that signals the hearing aid to provide the next incremental hearing correction. '999 patent, Ex. 1001 at 10:32-52; Fig. 3. The trigger can be generated a number of different ways, including once a number of clock cycles exceeds a preset limit, based on a calendar, or after a period of time programmed by the hearing aid user or preset by an audiologist. *Id*. at 10:36-52; *see also id*. at 7:51-54 (hearing aid user can program time period using an input interface of a computing device). The trigger can also be initiated by the hearing aid user. *Id*. at 10:47-53. For example, the hearing aid user can interact with a graphical user interface (GUI) through the input interface to cause a computing device to send the trigger so the hearing aid applies the next incremental hearing correction (*i.e.*, the next hearing correction filter). *Id*. at 8:3-16.

44. The progressive application of "filtered" hearing correction profiles continues until the user's full hearing aid profile is implemented. At that time, the hearing aid processor may provide an audible alert, such as through the speaker of the hearing aid, indicating to the hearing aid user that the adjustment process is complete. '999 patent, Ex. 1001 at 10:53-61.

IX. CLAIM CONSTRUCTION

45. In proceedings before the USPTO, I understand that the claims of an unexpired patent are given their broadest reasonable interpretation in view of the specification from the perspective of one skilled in the art. I have been informed that the '999 patent has not expired, nor will it during the pendency of this proceeding. In comparing the claims of the '999 patent to the known prior art, I have carefully considered the '999 patent and the '999 patent file history based upon my experience and my knowledge in the relevant field. In my opinion, the broadest reasonable interpretation of the claim terms of the '999 patent is generally consistent with an ordinary and customary meaning of those terms, as one skilled in the relevant field would have understood them.

46. I reviewed the claims of the '999 patent and in my opinion, certain claim terms have the following meanings based on the specification and an ordinary and customary meaning as understood by a POSA before April 13, 2010. The support for these meanings is further described in the following sections.

Claim Term	Proposed Claim Construction
hearing aid profile	a collection of acoustic configuration settings for a hearing aid which are used by a processor to shape acoustic signals to correct for a user's hearing loss
hearing correction filter	an adjustment applied by a processor to a hearing aid profile to reduce the level of correction provided to a user by application of the hearing aid profile
incremental hearing correction	a collection of acoustic configuration settings for a hearing aid which are used by a processor to shape

	acoustic signals to correct for a user's hearing loss, the collection of acoustic configuration settings representing an intermediate hearing adjustment to provide a modulated output signal having a level that is within a range between an uncompensated output level and the desired output level
incremental hearing correction filter	a hearing correction filter applied to provide a modulated output signal having a level that is within a range between an uncompensated output level and the desired output level

A. "hearing aid profile" (claims 1, 6, 9, 10, 16, 17, 20)

47. Independent claims 1, 6, and 10 (the only independent claims), as well as dependent claims 9, 16, 17, and 20, use the term "hearing aid profile." '999 patent, Ex. 1001. Based on my reading of the '999 patent, it is my opinion that the Patent Owner provided a definition for the term "hearing aid profile." Specifically, the '999 patent states that: "As used herein, the term 'hearing aid profile' refers to a collection of acoustic configuration settings for a hearing aid, such as hearing aid 202 depicted in FIG. 2, which are used by a processor 210 (in FIG. 2) to shape acoustic signals to correct for a user's hearing loss." '999 patent, Ex. 1001 at 2:40-44. The specification also confirms that a "hearing aid profile" is applied to signals to correct for a user's hearing loss. *Id.* at 2:46-49 ("Each of the hearing aid profiles are designed to compensate for the hearing loss of the user based on the user's particular hearing characteristics (impairment)."), 2:49-58.

48. It is my opinion that the ’999 patent’s definition of “hearing aid profile” is consistent with the term’s plain and ordinary meaning to a POSA. Accordingly, a POSA would have understood the term “hearing aid profile” to mean — a collection of acoustic configuration settings for a hearing aid which are used by a processor to shape acoustic signals to correct for a user’s hearing loss.

B. “hearing correction filter” (claims 1-6, 8-17)

49. Independent claims 1, 6, and 10 (the only independent claims), as well as dependent claims 2-5, 8, 9, and 11-18, use the term “hearing correction filter.” ’999 patent, Ex. 1001. Based on my reading of the ’999 patent and file history, it is my opinion that the Patent Owner provided a definition for the term “hearing correction filter.” Specifically, the ’999 patent states:

> “As used herein, the term “hearing correction filter” refers to a collection of filters for hearing aid 202, which are applied by processor 210 within hearing aid 202 to a hearing aid profile to reduce the level of correction provided to the user by application of the hearing aid profile. The collection of hearing correction filters may include a series of hearing correction adjustments designed to be applied in a sequence over a period of time to provide incremental corrections for the user’s hearing loss to ease the user’s transition from uncompensated to corrected hearing.” ’999 patent, Ex. 1001, 2:65-3:7.

50. Thus, the specification explicitly defines a "hearing correction filter" as "applied by [a] processor . . . to a hearing aid profile to reduce the level of correction provided to [a] user by application of the hearing aid profile" and makes clear that a "hearing correction filter" is an "adjustment[]" that is "applied" to the hearing aid profile. *Id.* The specification and claims further confirm this understanding of this term. '999 patent, Ex. 1001 at 3:7-10 ("a first hearing correction filter attenuates the hearing aid profile by a pre-determined amount, limiting the adjustment provided by hearing aid"), 3:10-15 ("[e]ach of subsequent hearing correction filter in the sequence increases the correction provided by . . . the hearing aid profile . . . until . . . the hearing aid profile is fully applied to provide the desired hearing correction for the user"), 3:32-36 ("incremental hearing corrections can be formed by applying one or more hearing correction filters to a selected hearing aid profile"), 4:15-22 ("a plurality of intermediate hearing sensitivity levels that fall within a range between hearing loss line 106 and hearing aid profile correction line 108, which intermediate hearing sensitivity levels are achieved by applying hearing correction filters to the selected hearing aid profile"), 4:22-27 ("[e]ach of the intermediate hearing sensitivity lines 110, 112, 114, 116, and 118 represents one or more adjustments to enhance the user's hearing sensitivity by applying an incremental hearing filter the selected hearing aid profile, reducing its hearing correction by a predetermined amount"), 5:31-38,

6:44-52, 8:55-67, 13:19-23, 13:26-31. Although one may argue that the language of claim 1 suggests that the "hearing correction filter" is applied to modulate an audio signal that has already been modulated by the hearing aid profile (*e.g.*, based on the language – "apply [a first hearing correction filter]… *to the modulated electrical signals*"), such an interpretation would directly contradict the disclosed embodiment(s) and the explicit definition of "hearing correction filter," which refers to the hearing correction filter being applied to a hearing aid profile, not an electrical signal that has already been modulated by such a hearing aid profile.

51. It is my opinion that the '999 patent's definition of "hearing correction filter" is consistent with the term's plain and ordinary meaning to a POSA. Accordingly, a POSA would have understood the term "hearing correction filter" to mean — an adjustment applied by a processor to a hearing aid profile to reduce the level of correction provided to a user by application of the hearing aid profile.

C. "incremental hearing correction" (claims 1-5, 10-13, 15-17)

52. Independent claims 1 and 10, as well as dependent claims 2 - 5, 11 - 13, and 15 - 17, use the term "incremental hearing correction." '999 patent, Ex. 1001. Based on my reading of the '999 patent and file history, it is my opinion that the Patent Owner provided a definition for the term "incremental hearing correction." Specifically, the '999 patent states:

> "As used herein, the term 'incremental hearing correction' refers to a collection of acoustic configuration settings for hearing aid 202 (such as a hearing aid profile described above), which are used by processor 210 within hearing aid 202 to shape acoustic signals to correct for a user's hearing loss. Each of the incremental hearing corrections represents an intermediate hearing adjustment to provide a modulated output signal having a level that is within a range between an uncompensated output level and the desired output level." '999 patent, Ex. 1001, 3:24-29.

53. The specification further confirms this definition. '999 patent at Abstract ("The processor is configured to apply a selected one of a sequence of incremental hearing corrections to the electrical signal to produce a modulated output signal to at least partially compensate for a hearing impairment of a user."), 3:32-56, Fig. 1, Fig. 3, 4:15-34, 5:3-14, 5:15-22, 5:31-33, 6:42-7:36, 10:10-25, 11:13-19, 11:41-48, 11:60-12:2, 12:3-10.

54. It is my opinion that the '999 patent's definition of "incremental hearing correction" is consistent with the term's plain and ordinary meaning to a POSA. Accordingly, a POSA would have understood the term "incremental hearing correction" to mean — a collection of acoustic configuration settings for a hearing aid which are used by a processor to shape acoustic signals to correct for a user's hearing loss, the collection of acoustic configuration settings representing an

intermediate hearing adjustment to provide a modulated output signal having a level that is within a range between an uncompensated output level and the desired output level.

D. "incremental hearing correction filter" (claims 1-5, 10-13, 15-17)

55. Independent claims 1 and 10, as well as dependent claims 2 - 5, 11 - 13, and 15 - 17, use the term "incremental hearing correction filter." '999 patent, Ex. 1001.

56. Claim 1 includes the limitations, "…apply a first one of a sequence of *incremental hearing correction filters* to the modulated electrical signals to produce a modulated output signal to reduce the amplitude of the modulated electrical signals produced by the selected hearing aid profile to a first level that is less than a level to compensate for the hearing impairment of the user; select a second one of the sequence of *incremental hearing correction filters* in response to receiving a trigger, the second one being designated to follow the first one in the sequence of *incremental hearing correction filters* and to reduce the amplitude of the modulated electrical signals produced by the selected hearing aid profile to a second level that is greater than the first level and less than the level to compensate for the hearing impairment of the user; and cause the speaker to output an alert when a final one of the sequence of *incremental hearing correction filters* is being

applied, the final one being the last hearing correction filter of the sequence of *incremental hearing correction filters*." '999 patent, Ex. 1001.

57. Claim 10 includes the limitations, "generate a sequence of *incremental hearing correction filters* based at least in part on a magnitude of a difference between a hearing aid profile and a hearing loss level associated with a user of the hearing aid, the sequence of *incremental hearing correction filters* including at least a first hearing correction filter and a second hearing correction filter; provide a first signal related to the first hearing correction filter of the sequence of *incremental hearing correction filters* to the hearing aid through the communication channel; and provide a second signal related to a second hearing correction filter of the sequence of *incremental hearing correction filters* to the hearing aid in response to receiving a selection of the second hearing correction filter from a user of the hearing aid." '999 patent, Ex. 1001.

58. The specification discloses three ways to provide "incremental hearing corrections"—they can be: (1) formed by applying one or more hearing correction filters to a selected hearing aid profile to produce the intermediate hearing aid profiles; (2) programmed by an audiologist; or (3) calculated dynamically as a function of a difference in decibels between the uncompensated level and the desired output level. '999 patent, Ex. 1001 at 3:32-41. As described in the specification, an "incremental hearing correction filter" is used in at least the

first way for providing an incremental hearing correction; namely an "incremental hearing correction filter" is a hearing correction filter applied to achieve an incremental hearing correction. *Id*. at 6:44-52 ("processor 210 selectively ***applies each a selected one of the hearing correction filters*** 220 ***to the selected hearing aid profile*** 218 ***to provide an incremental hearing correction*** for a period of time before advancing to ***a next incremental hearing correction by applying a next hearing correction filter*** 220 in a sequence"), 8:55-58 ("***incremental hearing corrections*** 224 and 276 are generated by ***applying hearing correction filters*** 222 and 274 ***to a selected one of hearing aid profiles*** 218 or 270, respectively"), 9:31-41 ("processor 210 selectively ***applies a selected one of the incremental hearing correction filters to the selected hearing aid profile*** for a period of time before advancing to a next incremental hearing correction filter in the sequence, providing incremental hearing adjustments from the uncompensated baseline to the fully-compensated hearing experience provided by the unfiltered hearing aid profile").

59. As explained above, and defined in the specification, an "incremental hearing correction" should be construed as "a collection of acoustic configuration settings for a hearing aid which are used by a processor to shape acoustic signals to correct for a user''s hearing loss, the collection of acoustic configuration settings representing an intermediate hearing adjustment ***to provide a modulated output signal having a level that is within a range between an uncompensated output***

level and the desired output level." Thus, the specification and construction discussed above confirm Petitioner's construction of this term as "a hearing correction filter applied to provide a modulated output signal having a level that is within a range between an uncompensated output level and the desired output level."

60. It is my opinion that the definition of "incremental hearing correction filter" is consistent with the term's plain and ordinary meaning to a POSA. Accordingly, a POSA would have understood the term "incremental hearing correction filter" to mean — a hearing correction filter applied to provide a modulated output signal having a level that is within a range between an uncompensated output level and the desired output level.

X. DISCUSSION OF RELEVANT PRIOR ART PATENTS AND PUBLICATIONS

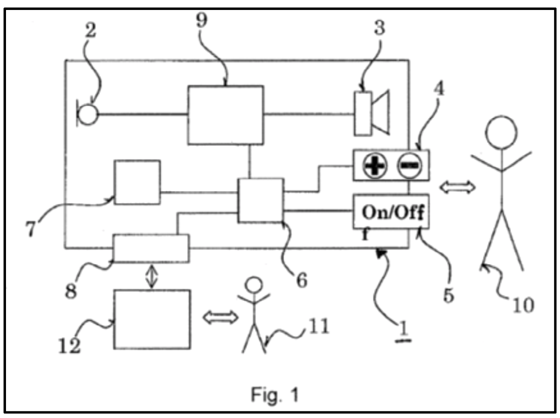
A. Fichtl

61. U.S. Patent No. 8,787,603 to Fichtl, et al. ("Fichtl") was filed on December 22, 2009 and issued on July 22, 2014. Fichtl, Ex. 1003. I have been informed that Fichtl is prior art under 35 U.S.C. § 102(e). Fichtl discloses a hearing aid with programming for automatic acclimatization. Fichtl, Ex. 1003 at Abstract.

62. When a hearing aid user is treated by an audiologist, specific audio processing parameters ("APPs") are chosen based on the user's hearing loss.

Fichtl, Ex. 1003 at 3:27-41; Fig. 1. The APP is typically volume "but may also be something else, as, for example, treble or noise canceling." *Id*. at 3:44-47.

63. The APPs are input by an audiologist (Fichtl, Ex. 1003 at Fig. 1, ref. no. 11) using a fitting interface (*id.* at Fig. 1, ref. no. 8; 13:36-37). The fitting interface (8) is coupled to the controller (6). *Id*. at Fig. 1. The fitting interface (8) communicates with a computing device (12), which is operated by the audiologist (11). *Id*. at Fig. 1, 3:35-48. The communication is represented by the bi-directional arrows between elements 8 and 12 in Fig. 1. *Id*.



Fichtl, Ex. 1003, Figure 1

64. In addition to the APPs, which constitute the user's hearing aid profile, the audiologist can input and apply a program for acclimatization to allow the user's hearing aid to incrementally adjust from less than the optimal APPs to an implementation of the full hearing aid profile. Fichtl, Ex. 1003 at 2:13-22, 2:40-51, 3:35-4:53. The program for acclimatization may adjust one or several audio parameters, such as volume, treble, or noise canceling. *Id*. at 3:42-48. However,

the hearing aid user can affect the programming, such as by increasing the speed at which the acclimatization occurs. *Id*. at Abstract.

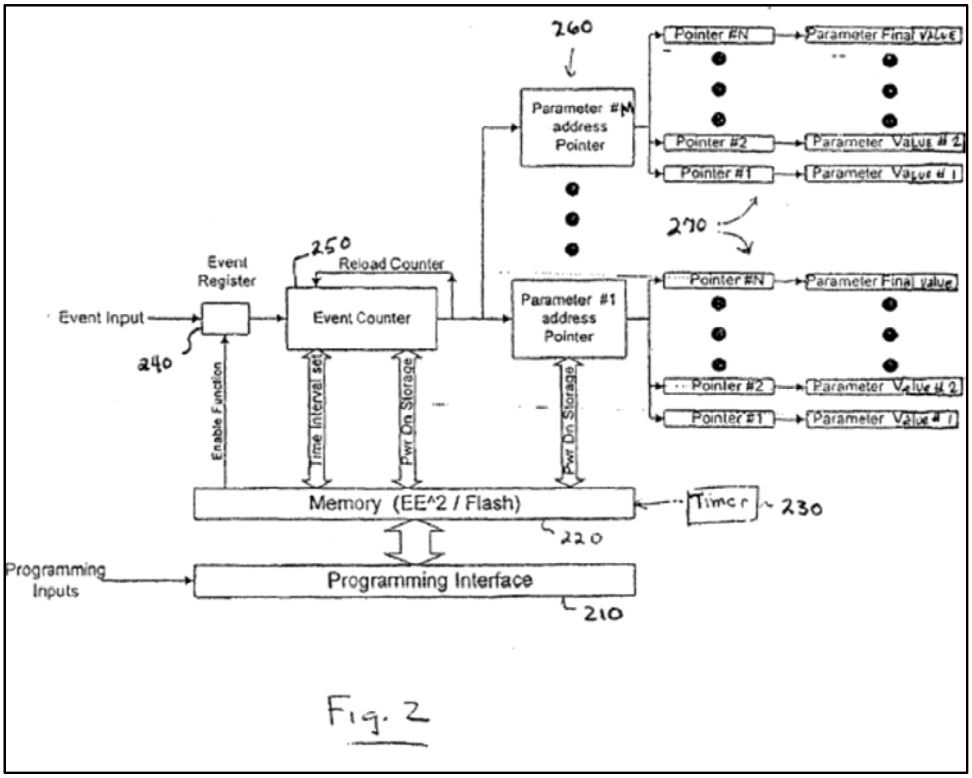
65. The acclimatization can be automated to progress based on different triggers, including those set by the audiologist as well as hearing aid user controlled parameters. For example, an audiologist may set the automatic acclimatization to progress stepwise upon a trigger. Fichtl, Ex. 1003 at 5:11-41. Additionally, the hearing aid user can adjust the rate of acclimatization. *Id*. at 3:35-4:53, 5:42-6:18.

B. Sacha

66. U.S. Patent Application Publication 2003/0215105 to Sacha was filed on May 16, 2002 and published on November 20, 2003 ("Sacha"). Sacha, Ex. 1004. I have been informed that Sacha is prior art under 35 U.S.C. § 102(b).

67. Sacha discloses a hearing aid that applies a gradual progression from an initial sub-optimal degree of compensation to the optimal level of compensation. Sacha, Ex. 1004 at Abstract, [0003], [0004]. The need for such a hearing aid is based on the recognition that a first-time hearing aid user may not initially tolerate a hearing aid that has been programmed to provide optimal hearing compensation. *Id*. at [0003], [0012]. Sacha notes the common practice of repeated fittings performed by an audiologist can be inconvenient and adds to the overall expense of a hearing aid. *Id*.

68. The hearing aid disclosed by Sacha may be programmed with a parameter set, where the parameter set consists of M parameters, each with a separate table that contains N alternative parameter values. Sacha, Ex. 1004 at [0012], [0013]; *see also* Fig. 2 (shown below).



Sacha, Ex. 1004, Figure 2

69. The parameters may dictate the filtering and amplification characteristics of the hearing aid or define other operating characteristics, such as the degree of compression or noise reduction. Sacha, Ex. 1004 at [0004], [0011].

C. Janssen

70. U.S. Patent Application Publication 2005/0036637 to Janssen ("Janssen") was filed on May 11, 2004 and published on February 17, 2005.

Janssen, Ex. 1005. I have been informed that Janssen is prior art under 35 U.S.C. § 102(b).

71. Janssen discloses a hearing aid that autonomously performs stepwise adjustment of one or more of the stored signal processing parameters from a starting point to an end point. Janssen, Ex. 1005 at Abstract, Fig. 1.

72. Janssen recognizes the problem of acclimating to hearing aid use, especially when the hearing aid user has been unable to hear certain frequencies, levels, and/or sound pressures for some time. Janssen, Ex. 1005 at [0002], [0003]. The user may find using an optimally adjusted hearing aid to be unpleasant. *Id.*

73. Janssen recognizes the advantages of gradually adjusting the operational parameters of a hearing aid automatically over time. Janssen, Ex. 1005 at [0004]-[0007].

D. Bisgaard

74. U.S. Patent No. 6,741,712 to Bisgaard was filed on June 21, 2001 ("Bisgaard") as a continuation of application no. PCT/DK99/00687, which was filed on December 8, 1999, and issued on May 25, 2004. Bisgaard, Ex. 1006. I have been informed that Bisgaard is prior art under 35 U.S.C. § 102(b).

75. Bisgaard discloses programmable hearing aids with data logging capabilities that may be used with a subscription agreement between a hearing aid user and a hearing aid dealer. Bisgaard, Ex. 1006 at 1:33-67. Bisgaard goes on to

describe the use of subscription services with hearing aids, similar to subscriptions for contact lenses, where an initial fee is paid when the agreement begins and then regular subscription fees are paid which cover an annual eyesight check-up and continuous deliveries of new lenses. *Id*. at 1:32-38. The hearing aid disclosed by Bisgaard can be used for such subscription services because it is possible to stop the current services if the hearing aid user does not pay the subscription fee, i.e., the hearing aid will stop working. *Id*. at 1:50-2:9.

76. An example hearing aid is described that includes a sound transducer, an analogue-digital converter, a digital processing and adjustment circuit for the processing of digital signals, corresponding to audio signals which are received by the transducer, memory units for the storage of data and programs for the digital processing and adjustment circuit, a digital-analogue converter and a speaker. Bisgaard, Ex. 1006 at 1:5-11. Bisgaard notes that such an example hearing aid is described in U.S. Patent No. 4,972,487 to Mangold. *Id*. at 1:12-15.

77. Bisgaard discusses how hearing aids can include different program settings, or profiles, for compensating for a user's hearing impairment. Bisgaard, Ex. 1006 at 1:12-20. These settings can be provided to or updated on the hearing aid using a PC. *Id*. at 2:31-39, 4:3-10, 6:19-26.

78. An audiologist or dealer will fit the user of a hearing aid and provide the hearing aid with programs and data that are necessary for its operation.

Bisgaard, Ex. 1006 at 6:33-44. The programs and data control the signal adjustment to compensate for the user's hearing loss. *Id*.

79. The hearing aid disclosed by Bisgaard can include programming for a "habituation system." Bisgaard, Ex. 1006 at 2:48-56; *see also id*. at 6:39-44. A habituation system is the same as an acclimatization program. "With such a habituation system, there will occur a gradual transition from no hearing aid to full compensation for the user's loss of hearing." *Id*. The acclimatization system allows a gradual transition from no hearing compensation to full compensation. *Id*.

80. The hearing aid can be programmed to include: (1) the maximum utilization time for a first subscription period, and (2) the acclimatization data or program for the following period, if an acclimatization system is used. Bisgaard, Ex. 1006 at 6:33-44. The following period could be a second subscription period or even the final subscription period for which the acclimatization system is in effect. *Id*. Upon being turned on, the hearing aid checks to see if the stored maximum utilization time is exceeded. *Id*. at 45-50.

81. One example hearing aid includes a counter element and a storage unit to track its utilization time, which is then compared to one or more maximum utilization time limit values. Bisgaard, Ex. 1006 at 1:55-67. When one or more of the limit values is reached, a special function is initiated. *Id*. The special function can be an alarm arrangement that alerts the hearing aid user by emitting an audible

alert in the form of a short acoustic signal. *Id*. at 2:15-22. The hearing aid user is alerted when the first limit value is reached. *Id*. In addition to the alert, reaching a limit value can initiate a change in the parameters or programs that influence operation of the hearing aid, such as discontinuing the hearing aid's function of signal adjustment. *Id*. at 2:23-30.

82. Bisgaard teaches that when a habituation system (acclimatization program) is used, the parameters or programs used for the signal processing can be changed, and such changes may occur after a limit value is reached. Bisgaard, Ex. 1006 at 7:4-9. For example, after a limit value is reached, the new acclimatization data can be selected from the electronically erasable programmable read-only memory. *Id*.; *see also* 3:66-4:6.

83. The audible alert may consist of an alarm arrangement, whereby in the event of a first limit value being exceeded, the hearing aid user is made aware that the subscription period has expired or is nearing expiry, *e.g.*, by the hearing aid emitting a short acoustic signal which is possibly repeated. Bisgaard, Ex. 1006 at 2:15-22. The short acoustic signal can be output by the hearing aid's speaker. If the subscription is not renewed, the audible alarm can be combined with the deactivation of the hearing aid after a second limit value is exceeded. *Id*.

84. If a maximum utilization time limit value is met and the current subscription period is expired, an audio alarm can be output to the hearing aid user

through the hearing aid's speaker. Bisgaard, Ex. 1006 at 45-57. After the alarm, the hearing aid no longer uses the normal operating program. *Id.* at 6:53-57 ("after which the normal operating programme will not be initiated"). If the maximum utilization time limit value is not met, then the hearing aid will continue executing its operating program and functioning in the normal manner. *Id*. at 6:58-61.

85. If the subscription period has expired and the hearing aid is deactivated, the hearing aid can be reactivated when it detects a series of reactivation signals. Bisgaard, Ex. 1006 at 6:62-7:12. When the hearing aid is reactivated, the hearing aid may implement changed parameters or a changed program. *Id.*; *see also* claim 8 ("The hearing aid according to claim 5, wherein upon detection of a reactivation signal, a changing of parameters and/or programs is also carried out.").

E. Mangold

86. U.S. Patent No. 4,972,487 ("Mangold") was filed on May 16, 1989 as a continuation of U.S. Application No. 07/175,233 filed March 30, 1988. Mangold, Ex. 1007. Mangold issued on November 20, 1990. I have been informed that Mangold is prior art under 35 U.S.C. § 102(b).

87. Mangold is directed to "auditory prostheses including hearing aids." Mangold, Ex. 1007 at 1:9-15. Mangold discloses a plurality of "programs" that include a collection of parameters and/or coefficients in a hearing aid program for

processing audio signals. *Id*. at 2:28-33, 4:47-58, 1:17-22, 1:23-29. During an audiological fitting, the programs are stored in memory and adjusted to correct for the hearing loss of a particular hearing aid user. *Id*. at Abstract, 2:3-11, 1:17-22. Mangold discloses that a program may be selected by a user or in response to the current sound environment of the hearing aid user. *Id*. at 3:13-15 ("A manual program control switch 18 is provided for the user of the device to select from among the several programming options stored in memory 16"), 1:40-49, 1:17-22.

88. Mangold discloses a hearing aid in communication with a remote control that includes a control unit, a transmitter, a manual program control, and, optionally, a memory. Mangold, Ex. 1007 at 1:59-2:2, 3:38-59, Fig. 3, 3:10-15, Fig. 5, 4:11-26. The hearing aid user can select a stored program using the manual program control 24, which is part of the remote control. *Id*. at 3:13-15, 3:38-43, Fig. 3, 4:11-16, Fig. 4. The selected program is stored in the memory (*Id*. at 3:10-13, Fig. 1), which can be located in the hearing aid or in the remote control unit. *Id*. at 3:27-29, Fig. 2, 4:11-21, Fig. 5. The selected program can be identified by a program number. *Id*. at 3:49-59 (describing transmitting a "program number" identifying a selected program to the hearing aid when the memory is located in the hearing aid).

F. DE961

89. German patent publication DE19542961 ("DE961") was published on May 15, 1997. DE961, Ex. 1009. I have been informed that DE961 is prior art under 35 U.S.C. § 102(b).

90. DE961 discloses a hearing aid with, *inter alia*, a control unit, a memory system, a starting setting for operating parameters, a target setting for operating parameters, and a timer. DE961, Ex. 1009 at 2-3.

91. The hearing aid may include an acclimatization program that executes interim levels of adjustment over a period of time. DE961, Ex. 1009 at Abstract ("an adjustment of the operating parameter according to the setting of the starting situation to the operating parameter of the target situation can be carried out over a specific time interval"); *see also id*. at 3 ("operating parameters of the starting situation should be adjusted so that the patient senses the sound to be pleasant … by way of the method according to the invention … the sound will be brought gradually to the optimal state"). The purpose of the acclimatization program is to reduce discomfort for the hearing aid user. *Id*. at Abstract, 2.

92. DE961 references a prior art disclosure of a circuit with a control unit for a hearing aid, which, for each channel, makes the adjustment of " an adaptable amplification for selective enhancement of main signals and suppression of background noise" possible. DE961, Ex. 1009 at 2 (citing U.S. Patent No.

5,396,560). It also refers to a circuit for a hearing aid where a channel output or starting signal is compared with a threshold value of the respective channel depending on the fitting of the hearing aid to the user. *Id*. (citing WO 94/23548).

93. The hearing aid disclosed in DE961 includes an initial setting for operating parameters ("operating parameter setting of the starting situation") and a target setting for operating parameters that are stored in memory. DE961, Ex. 1009 at 2-3. The operating parameters used by the hearing aid are adjusted from the initial parameter setting to the target setting over a specific time interval. *Id.* This adjustment can occur in uniform steps in an order of magnitude or in time intervals. *Id*. at 3. For example, the acclimatization program may use a series of time constants, which is the time interval for which particular operating parameters are implemented on the hearing aid. *Id.* The hearing aid's control unit may include a timer, or other timing element, to keep track of the elapsed time and compare it to the time constants. *Id*.

94. The operating parameters to be adjusted, either individually or together, can include, "the amount of preamplification, the magnitude of the compression ratio, the magnitude of the threshold value, and/or the magnitude of the time constants of the compression circuit, the frequency bandwidth with which the sound can be adjusted to the individual hearing ability, and/or the maximum

output level." DE961, Ex. 1009 at 4. These operating parameters "are adjusted via the time constants of the timer 15 in a continuous manner." *Id*.

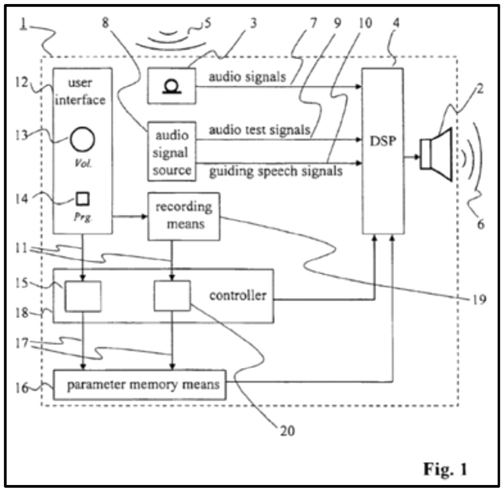
95. DE961 discloses that the hearing aid user, or patient, has the ability to control aspects of the acclimatization program. The patient can adjust the timing of adjustments for the operating parameters. DE961, Ex. 1009 at 3 ("the patient himself or herself can choose a longer time constant"); 4-5 ("It is possible with the timer 15 for the patient to preferably select alternatively a plurality of time constants"). For example, if the adjustment of the operating parameters in the acclimatization program occurs too rapidly for the patient, "the patient can select a slower time constant." *Id*. at 5. In one embodiment, the parameters can be adjusted "on an individual and patient-related basis by means of an adjustment program." *Id.* at 5.

G. Roeck

96. U.S. Patent No. 7,933,419 to Roeck et al. ("Roeck") was filed on October 5, 2005 and issued on April 26, 2011. Roeck, Ex. 1010. I have been informed that Roeck is prior art at least under 35 U.S.C. § 102(e).

97. Roeck teaches a hearing device, such as a hearing aid, that can be fitted or adjusted by the hearing aid user without the help of an audiologist. Roeck, Ex. 1010 at Abstract; 1:55-2:3. Roeck's hearing device can also include a user interface, a remote control, or a separate device from the hearing aid. *Id*. at 3:6-10,

3:30-38. The other device, such as the remote control or separate device from the hearing aid, may also be the audio source or may store the parameter settings. *Id.*; *see also id.* at Abstract, 6:40-45.



Roeck, Ex. 1010, Figure 1

98. The hearing aid user can fit the hearing aid to include initial settings, or the hearing aid can choose from preset initial settings, *e.g.*, an imprinted "1" or "2" or "3", wherein, e.g., in the case of a hearing aid, "1" could indicate an initial parameter setting for a user with light hearing loss, "2" could indicate an initial parameter setting for a user with moderate hearing loss, and "3" could indicate an initial parameter setting for a user with severe hearing loss. Roeck, Ex. 1010 at 4:40-56. The hearing aid can be audio-fitted using only the hearing device or, alternatively, the hearing device's audio-fitting may be part of a more extensive fitting that uses an external personal computer. *Id.* at 8:34-42.

99. The hearing aid user interface can include multiple controls, such as a volume wheel or a program change knob. Roeck, Ex. 1010 at 6:23-33. If a hearing aid user presses the program change knob, he can toggle between listening and fitting mode. *Id*. at 7:2-20. The hearing aid user can also use the program change knob to change from fitting a high frequency band to a mid-frequency band to a low-frequency band, and adjusting each frequency band as needed. *Id.*

100. There are a number of parameters that may be pre-configured or derived from the parameter settings obtained from the hearing aid user input, including "knee-point levels, knee-point gains, expansion slopes, compression slopes, maximum gain settings, maximum output values, and other parameters." Roeck, Ex. 1010 at 8:66-9:3.

101. Roeck teaches that the hearing aid user can directly audio-fit various parameters of his own hearing aid, as well as a maximum power output (MPO) that can be set by the hearing aid user either explicitly or implicitly. Roeck, Ex. 1010 at 8:58-65; Fig. 3. As shown in Fig. 3, the MPO acts as a maximum bar to certain hearing aid user adjustments, so the hearing aid user is not able to adjust the hearing aid to a point that would be uncomfortable. *Id.*

XI. CLAIMS 1-5 AND 16 ARE OBVIOUS OVER FICHTL (EX. 1003) IN VIEW OF MANGOLD (EX. 1007) AND BISGAARD (EX. 1006)

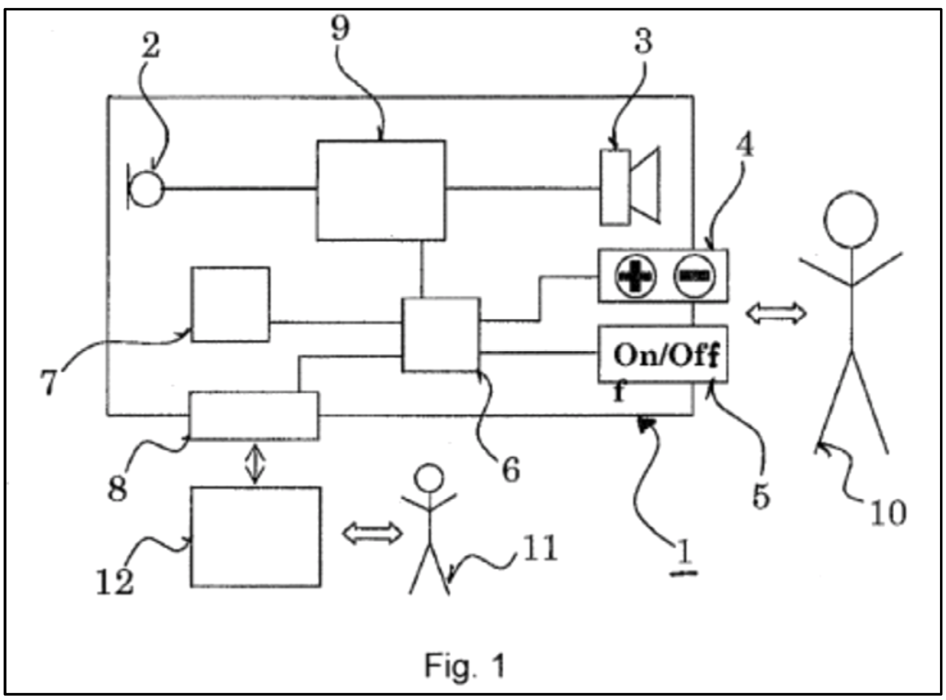
102. As provided in my detailed analysis below, it is my opinion that claims 1-5 and 16 of the '999 patent are rendered obvious by Fichtl in view of Mangold and Bisgaard.

A. [Claim 1. Preamble]

103. As I explained above, Fichtl discloses a "hearing aid." *See* Fichtl, Ex. 1003 at Abstract, 3:21-34, Fig 1, 13:30.

B. [Claim 1.1] - "a microphone to convert sound into electrical signals"

104. Fichtl discloses "a microphone to convert sound into electrical signals." Fichtl, Ex. 1003 at Fig. 1 (element 2), 3:23-24 ("Sounds are picked up by a microphone 2, processed by a signal processor 9 …"), 13:31.



Fichtl, Ex. 1003, Figure 1

A. [Claim 1.2] - "a speaker to output audible sound"

105. Fichtl discloses "a speaker to output audible sound." Fichtl, Ex. 1003 at Fig. 1 (element 3), 3:23-25 ("Sounds … are presented to a hearing device user 10 by a receiver 3. The magnitude of the amplification can be controlled by a volume control 4."), 13:34-35. The receiver of Fichtl outputs audible sound to the hearing aid user. *Id.*

B. [Claim 1.3] - "a processor"

106. Fichtl discloses "a processor." Fichtl, Ex. 1003 at Fig. 1, 3:23-34 ("Sounds are … processed by a signal processor 9 …. The signal processing is based on audio parameters. A controller 6 is adapted to set such parameters …. The controller 6 is adapted to execute an acclimatization algorithm of the kind described further down below"), 13:32-33, 13:44. Thus, controller 6 and/or signal processor 9 correspond to the claimed "processor."

C. [Claim 1.4] - "a memory to store instructions"

107. Fichtl discloses "a memory to store instructions." Fichtl, Ex. 1003 at Fig. 1, 2:19-27 ("a non-volatile memory"), 3:23-34 ("Sounds are … processed by a signal processor 9 …. The signal processing is based on audio parameters. A controller 6 is adapted to set such parameters …. There is a non-volatile memory 7 to store parameters while the hearing device 1 is switched off. The controller 6 is adapted to execute an acclimatization algorithm of the kind described further down

below"); 4:25-51 ("the acclimatization program is controlled by software being executed on the controller 6 …"), 13:43. Because the controller executes the acclimatization algorithm through software and the signal processor processes sounds, a POSA would have understood that memory 7 (or a separate memory) stores instructions for executing the acclimatization program.

108. Additionally, Mangold's hearing aid includes a memory to store instructions for operating the hearing aid. Mangold, Ex. 1007 at 3:10-13 ("A programmable memory with logic 16 stores a plurality of programs for controlling the signal processor 12 in operating on signals from microphone 10"), 2:67-3:2, Fig. 1. One example hearing aid includes a microphone for picking up sound and converting it to an electrical signal, a signal processor and associated slave memory for operating on the electrical signal generated by microphone 10 in accordance with one of a plurality of signal-processing programs, and a speaker for audibly transmitting the processed signals, in addition to the memory. *Id.* at 3:2-13.

109. A POSA would recognize that the instructions executed by the hearing aid disclosed by Fichtl likewise are stored in "a memory to store instructions" within the hearing aid. It also would have been obvious, and within the knowledge of the POSA, to modify Fichtl in light of the disclosure of Mangold to include a memory to store instructions for operating the hearing aid.

D. [Claim 1.5.1] – "…instructions, which when executed by the processor, cause the processor to: receive a selection of a hearing aid profile from a plurality of hearing aid profiles, the selected hearing aid profile configured to modulate the electrical signals to a level to compensate for a hearing impairment of a user"

110. The combination of Fichtl and Mangold results in "instructions, which when executed by the processor, cause the processor to: receive a selection of a hearing aid profile from a plurality of hearing aid profiles, the selected hearing aid profile configured to modulate the electrical signals to a level to compensate for a hearing impairment of a user."

111. Fichtl discloses fitting a hearing aid to a hearing aid user. Fichtl, Ex. 1003 at 3:37-39 ("The hearing device 1 is initially fitted to a hearing loss of a hearing device user 10 . . .").

112. As explained above, Fichtl discloses that during the audiological fitting, various APPs, such as "volume," "treble or noise canceling" are chosen for the user's hearing loss. Fichtl, Ex. 1003 at 3:27-48, Abstract ("An initial power-on value (iPOV) and a target power-on value (tPOV), which is to be reached at the end (H) of the acclimatization phase, may be programmed by an audiologist."), Fig. 1 (depicting fitting interface (8) that communicates with computing device (12) operated by audiologist (11)), 13:36-37 ("a fitting interface (8) for adjusting said hearing device (1) to the needs of said hearing device user (10)"), 13:45-14:1 ("fitting interface (8) is adapted to write an initial power-on value (iPOV) and a

value indicative of a target power-on value (tPOV) for said audio processing parameter (APP) to said non-volatile memory (7)").

113. These APPs constitute a collection of acoustic configuration settings for a hearing aid which are used by Fichtl's signal processor 9 to shape acoustic signals to correct a user's hearing loss and thus correspond to the claimed "hearing aid profile.". *Id.* at 3:23-28 ("Sounds are picked up by a microphone 2, processed by a signal processor 9 and are presented to a hearing device user 10 . . . . The signal processing is based on audio processing parameters."), Fig. 1.

114. A POSA would have understood that during an audiological fitting, multiple such hearing aid profiles are programmed into the memory of the hearing aid. The hearing aid's processor receives a selected profile to compensate for the hearing impairment of the user during use.

115. Mangold is directed to "auditory prostheses including hearing aids." Mangold, Ex. 1007 at 1:9-15. Mangold discloses a plurality of "programs" that include a collection of parameters and/or coefficients in a hearing aid program for processing audio signals. *Id.* at 2:28-33, 4:47-58, 1:17-22, 1:23-29. During an audiological fitting, the programs are stored in memory and adjusted to correct for the hearing loss of a particular hearing aid user. *Id.* at Abstract, 2:3-11, 1:17-22. Mangold discloses that a program may be selected by a user or in response to the current sound environment of the hearing aid user. *Id.* at 3:13-15 ("A manual

program control switch 18 is provided for the user of the device to select from among the several programming options stored in memory 16"), 1:40-49, 1:17-22. Thus, Mangold discloses selecting a hearing aid profile from a plurality of hearing aid profiles, the selected hearing aid profile configured to modulate the electrical signals to a level to compensate for a hearing impairment of a user.

116. The combination of Fichtl and Mangold renders obvious the selection of a hearing aid profile from a plurality of hearing aid profiles as provided in claim 1. It would have been obvious for Fichtl's memory 7 for storing APP values to store a plurality of hearing aid profiles and Fichtl's controller 6 to receive a selection of a stored hearing aid profile from the plurality of hearing aid profiles. The selection of a hearing aid profile could be, for example, in response to the current sound environment of the hearing aid user, as taught by Mangold. Mangold, Ex. 1007 at 1:40-49 (disclosing "environmentally selected events, such as selection of settings, parameters, or algorithms, where such selection is based on an automatic computation in response to the current sound environment of the wearer"), 3:49-66, Fig. 3. Providing a plurality of hearing aid profiles would allow a profile to be selected based on the hearing aid user's current sound environment to better compensate for hearing loss in that environment.

E. [Claim 1.5.2] "apply a first one of a sequence of incremental hearing correction filters to the modulated electrical signals to produce a modulated output signal to reduce the amplitude of the modulated electrical signals produced by the selected hearing aid

profile to a first level that is less than a level to compensate for the hearing impairment of the user"

117. Fichtl discloses a hearing aid with a processor that executes stored instructions to "apply a first one of a sequence of incremental hearing correction filters to the modulated electrical signals to produce a modulated output signal to reduce the amplitude of the modulated electrical signals produced by the selected hearing aid profile to a first level that is less than a level to compensate for the hearing impairment of the user."

118. Fichtl's controller is programmed to execute an acclimatization algorithm where the amount of compensation for the user's hearing loss increases automatically over time. Fichtl, Ex. 1003 at Abstract ("The intensity of the hearing device is increased in the long term"); 3:32-34 ("controller 6 is adapted to execute an acclimatization algorithm . . ."); 4:25-26 ("acclimatization process is controlled by software being executed on the controller 6"). As represented by the curve marked "X" on the graph depicted in Fig. 2, the acclimatization algorithm executed by controller 6 increases the value of an APP over time. *Id.* at Fig. 2, 3:35-36 ("FIG. 2 shows how an audio processing parameter APP is changed over time in a hearing device 1"); 3:42-4:15; 4:25-67. An intermediate value X is increased slowly while the hearing aid is on and held constant, stored in memory, while the hearing aid is off, such that each time the hearing aid is turned on the APP is set to

the last value for X as stored in memory. *Id.* at Fig. 2, 3:55-57, 3:66-4:7, 4:31-36, 4:41-53.

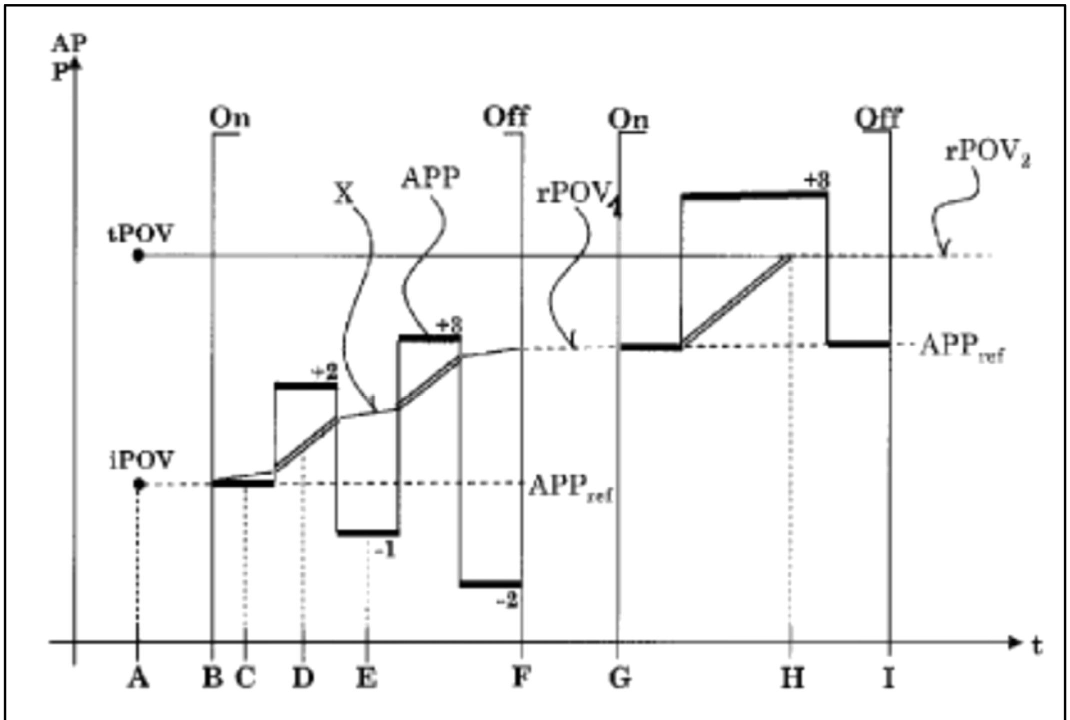


Fig. 2

Fichtl, Ex. 1003, Figure 2

119. As shown in Fig. 2, the APP starts at an initial power-on value (iPOV) selected to provide a smaller degree of compensation as compared to the target power-on value (tPOV) for the APP, which is the value for the APP corresponding to the selected hearing aid profile that compensates for the user's hearing impairment, and the compensation increases over time to a replacement power-on value (rPOV) each time the hearing aid us turned on until it reaches tPOV. Fichtl, Ex. 1003 at Fig. 2, 3:42-48 ("At time 'A,' a fitter programs an initial power-on

value iPOV for the audio processing parameter as well as a target power-on value tPOV…The target power-on value tPOV is, for example, 10 dB higher than the initial power-on value iPOV"), Abstract ("An initial power-on value (iPOV) and a target power-on value (tPOV), which is to be reached at the end (H) of the acclimatization phase, may be programmed by an audiologist."), 3:49-4:24, 4:25-67. Processor 9 uses APP values provided by controller 6, including the reduced values iPOV and rPOV relative to tPOV generated by the acclimatization algorithm, to process sounds for the hearing device user. *Id*. at Fig. 2; 3:23-34.

120. Fichtl's acclimatization algorithm corresponds to adjustments applied by controller 6 to the collection of APPs of processor 9 to reduce the level of correction provided to the hearing device user by application of the hearing aid profile. In other words, Fichtl's acclimatization algorithm as executed by controller 6 comprises a sequence of "hearing correction filters." Fichtl, Ex. 1003 at Fig. 2; 3:23-34.

121. The APP adjusted by the acclimatization algorithm may correspond to, for example, volume or treble. Fichtl, Ex. 1003 at 3:42-47. A volume APP corresponds to the loudness, or amplitude, of the output signal. *Id*. at 3:25-26 ("The magnitude of the amplification can be controlled by a volume control 4."), 3:44-48 ("The audio processing parameter APP is typically volume . . . . The target power-on value tPOV is, for example, 10 dB higher than the initial power-on value

iPOV.”), 6:42-48 (“The adjustments in the first adjustment direction are implemented by applying a faster learning speed . . . . If the audio processing parameter APP is volume, the first adjustment direction is louder . . .”). A treble APP corresponds to the loudness, or amplitude, of the higher frequencies. By applying an intermediate value that is lower than a target value, tPOV, for the volume or treble APP, Fichtl provides a modulated output signal having a level that is within a range between an uncompensated output level and the desired output level. Therefore, due to the produced modulated output signal, Fichtl’s hearing correction filters are “incremental hearing correction filters,” and the modulated output signal is produced as in the ’999 patent. *See* ’999 patent, Ex. 1001 at 3:32-41.

122. Fichtl’s hearing correction filters, corresponding to intermediate APP values, provide a modulated output signal with a reduced amplitude relative to the output signal that would be provided by Fichtl’s hearing aid profile, corresponding to the target APP value tPOV. Fichtl, Ex. 1003 at 3:35-4:24, Fig. 2. Applying the intermediate APP value serves to reduce the amplitude of the modulated signals to a level less than a level to fully compensate for the hearing impairment of the user, as recited in this claim 1 limitation. *Id.*

123. To the extent this limitation is interpreted contrary to the proper construction of the term “hearing correction filter” so it requires applying a

hearing correction filter to electrical signals already modulated by a hearing aid profile, rather than to a hearing aid profile as properly construed, a POSA would have understood that achieving a reduced APP, such as volume or treble, could be implemented in a number of equivalent ways to obtain an output signal corresponding to any of the intermediate values for the APP. The acclimatization takes an input signal I, and applies a target correction T reduced by R1, to obtain an output signal O. The acclimatization could be applied as an adjustment to the target power-on value that is applied either before or after applying the target correction (that is, apply R1 to the input signal and then T, or apply T to the input signal and then R1). Alternatively, the acclimatization could be applied by adjusting the target correction by the reduction amount, and applying that result to the input signal. In simple terms:

> O = (I*R1)*T [apply reduction first]
>
> O = (I*T)*R1 [apply reduction second, alternative claim interpretation]
>
> O = I*(T*R1) [create adjustment value as in Fichtl and proper claim interpretation]

The key is that the output signal O can be achieved by any of the three possible applications.

124. A POSA would have understood that the same resulting modulated output signal O could be achieved regardless of the order of application of incremental hearing correction filter, hearing correction filter, or hearing aid

profile as reflected in the three implementations above and regardless of whether the incremental hearing correction filters are previously stored or calculated dynamically. Thus, it would have been obvious to a POSA at the time of the invention to modify Fichtl to the alternative implementation since it would require no more than a routine, known implementation change, which yields the same, predictable result.

F. [Claim 1.5.3] "select a second one of the sequence of incremental hearing correction filters in response to receiving a trigger, the second one being designated to follow the first one in the sequence of incremental hearing correction filters and to reduce the amplitude of the modulated electrical signals produced by the selected hearing aid profile to a second level that is greater than the first level and less than the level to compensate for the hearing impairment of the user"

125. Fichtl discloses this part of claim 1. Fichtl, Ex. 1003 at Abstract ("The intensity of the hearing device is increased in the long term"), Fig. 2, 3:35-36 ("FIG. 2 shows how an audio processing parameter APP is changed over time in a hearing device 1"). Fichtl teaches an acclimatization algorithm that is executed by the controller to adjust the audio processor parameter settings incrementally over time. *Id.* at 3:35-36, 4:41-51 (describing the acclimatization algorithm to obtain intermediate values), 5:11-41 (describing the creation of progressive intermediate adjustment values); *see also id.* at 5:42-7:48 (describing additional variations for the creation of progressive intermediate adjustment values).

126. In one example disclosed by Fichtl, after a user first turns on the hearing aid and the iPOV is implemented, the acclimatization algorithm is executed and a second APP setting is stored as a replacement power-on-value rPOV when the hearing aid is turned off. Fichtl, Ex. 1003 at Fig. 2, 3:66-4:4; *see also id.* at 5:11-41. This second APP value rPOV corresponds to a second in a sequence of incremental hearing correction filters and is designated to follow the initial APP value iPOV, corresponding to a first in a sequence of incremental hearing correction filters. *Id.* at Fig. 2, 3:66-4:7, 4:31-36, 4:41-53. The second APP value is implemented in response to, and thus triggered by, the hearing aid being powered-on. *Id.* The sequence of incremental APP values in Fichtl can be "stored in said non-volatile memory" or "calculated from values stored in said non-volatile memory." *Id.* at 4:25-36. The APP adjusted by the acclimatization algorithm may correspond to volume which corresponds to the loudness, or amplitude, of the output signal. *Id.* at 3:25-26, 3:42-48, 6:42-48. The second APP value "is greater than the first level and less than the level to compensate for the hearing impairment of the user." *Id.* at Abstract ("The intensity of the hearing device is increased in the long term"), Figs. 2-4, 3:54-4:4 (describing how the intermediate value is increased as shown in Fig. 2), 5:11-41 (describing the generation of new incremental values, each a constant value greater than the prior

one, as shown in Fig. 3), 5:42-6:18 (describing the generation of new incremental values, each greater than the prior one, as shown in Fig. 4).

G. [Claim 1.6] "cause the speaker to output an alert when a final one of the sequence of incremental hearing correction filters is being applied, the final one being the last hearing correction filter of the sequence of incremental hearing correction filters"

127. An obvious modification of Fichtl in view of Bisgaard results in a hearing aid with instructions to cause the processor to "cause the speaker to output an alert when a final one of the sequence of incremental hearing correction filters is being applied, the final one being the last hearing correction filter of the sequence of incremental hearing correction filters." Fichtl, Ex. 1003; Bisgaard, Ex. 1006. Fichtl and Bisgaard each disclose hearing aids that can execute acclimatization programs. *Id.* The hearing aids include a speaker, as described above. *Id.*

128. Bisgaard discloses a hearing aid that when fitted is provided with programs for processing signals to compensate for the hearing aid user's hearing loss and a limit value corresponding to a subscription period. Bisgaard, Ex. 1006 at 6:33-44. The programs can effect an acclimatization system for "gradual transition from no hearing aid to full compensation for the user's loss of hearing." *Id.* at 2:48-56; *see also id.* at 6:33-44, 7:4-9. When the limit value is exceeded, the hearing aid can initiate changing of the programs and execute an alarm or acoustic signal to alert the hearing aid user. *Id.* at 2:1-9, 2:15-30, 6:45-57. Thus, Bisgaard

describes a system that will sound an alarm to alert the user when changing hearing aid programs for processing signals to compensate for hearing loss, including when a final one of the acclimatization sequence is applied.

129. It would have been obvious to modify Fichtl in view of Bisgaard to cause the speaker to output an alert when a final one of the sequence of incremental hearing correction filters is being applied, the final one being the last hearing correction filter of the sequence of incremental hearing correction filters as provided in claim 1. For example, Fichtl discloses an example in which the power on value "remains constant after the acclimatization phase ends." Fichtl, Ex. 1003 at 5:1-2. A POSA would have been motivated to modify Fichtl to incorporate the audible alarm of Bisgaard to alert the hearing aid user that the hearing aid is changing programs from an acclimatization algorithm, that has terminated, to a regular program, so the user knows that he or she should now be hearing at a fully compensated level and can alert their audiologist if the operation does not seem right.

H. A POSA Would Have Combined Fichtl, Mangold, and Bisgaard

130. Fichtl, Mangold, and Bisgaard all relate to hearing aid systems, which may include acclimatization programs. Fichtl, Ex. 1003; Mangold, Ex. 1007; Bisgaard, Ex. 1006.

131. According to Fichtl,

> Usually, it takes some time for a user to get used to a hearing device. This process is called acclimatization and may take e.g. from several weeks up to half a year. Typically, hearing devices are tuned by a specialist such as an audiologist. It has been shown that acclimatization can be made more comfortable for a user if the intensity of the hearing device is initially low and is increased gradually during an acclimatization phase until target intensity is reached. Practically, this means that the hearing device user has to return to the specialist several times for a retuning. At each visit the intensity of the hearing device is increased. In order to reduce the number of visits necessary and to make the adjustment more steady, it has been proposed to increase the intensity of hearing device automatically, a feature which is termed in this document "automatic acclimatization management".

Fichtl, Ex. 1003 at 1:19-34. Fichtl is focused on acclimatization. *E.g., id.* at Abstract, 2:13-60. Bisgaard's hearing aid system, with an audible alarm, can be used with an acclimatization ("habituation") system. Bisgaard, Ex. 1006 at 2:48-56, 6:33-44, 7:4-8. Thus, Fichtl and Bisgaard both address acclimatization to incrementally increase hearing correction to a desired value, and a POSA would have been motivated to look at and adapt the specific implementations in each. Doing so would involve nothing more than combining known elements in known ways to address known problems. Furthermore, Bisgaard discloses using its acclimatization system in the hearing aid of Mangold. *Id*. at 1:5-15; *see also* Mangold at 2:3-15. Thus, a POSA would have been motivated to incorporate acclimatization systems into the hearing aid of Mangold.

132. For example, to the extent not expressly disclosed in Fichtl, a POSA would have been motivated to modify the hearing aid of Fichtl to include "a memory to store instructions" as taught by Mangold, thereby resulting in a hearing aid that is equipped to store and to execute instructions for an automatic acclimatization program as described. The combination would have involved nothing more than combining known elements of a processor and memory to address a known problem of the need for more effective processing. For example, by including a memory, the hearing aid would not require a device other than the hearing device to store the instructions to operate.

133. A POSA would have recognized that the memory provided in the hearing aid of Mangold for storing executable code and data would be included in the hearing aid of Fichtl to store executable instructions and data. The combination would have required nothing more than combining memory and a hearing aid according to known methods, with no change in their respective functions, to yield predictable results.

134. A POSA would have recognized, as discussed in Mangold, that for many years hearing aids had included a plurality of parameter sets (profiles) from which one can be selected automatically or manually to provide compensation for the user's hearing impairment. Accordingly, a POSA would have expected Fichtl's hearing aid to select from a plurality of profiles or at least to be easily modified (in light of Mangold) to permit selection from a plurality of profiles, since Fichtl already discloses the use of a profile. Fichtl, Ex. 1003 at 3:37-48. This combination would have involved nothing more than combining prior art elements according to known methods, with no change in their respective functions, to yield predictable results.

135. A POSA would have recognized that a hearing aid user might need or want additional fitting after an automatic acclimatization program has completed and the "optimal" hearing correction has been implemented. For example, if the user's hearing loss worsened during the acclimatization period or if the user was otherwise unsatisfied with the resulting hearing compensation, it would be beneficial for the user to re-visit the audiologist for further fitting/adjustment. Therefore, a POSA would have had a reason to modify the hearing aid of Fichtl to include an audio alert, as taught by Bisgaard, upon the completion of an automatic acclimatization program in order to alert the user that the optimal hearing profile had been reached. A POSA would have recognized that such an audio alert would

have been beneficial to a hearing aid user to notify the user that the "optimal" level of hearing compensation has been reached. Then, the user would be able to assess satisfaction with hearing aid performance and, if needed, schedule a follow-up visit with the audiologist for adjustments. Therefore, outputting an audio alert when the final, "optimal" profile is applied would have been obvious to a POSA. To modify the hearing aid of Fichtl to include an audible alert, as taught by Bisgaard, to alert a user to completion of the acclimatization algorithm would have involved nothing more than a combination of prior art elements, including the hearing aid, speaker, processor, etc., according to known methods, with no change in their respective functions, to yield the predictable result that the hearing aid's speaker would output an audio alarm upon completion of the acclimatization program.

I. [Claim 2] "each of the incremental hearing correction filters comprises a collection of acoustic configuration settings configured to modulate the electrical signal to a level that is within a range between an uncompensated hearing level of the user and the level to compensate for the hearing impairment of the user"

136. An obvious modification of Fichtl in view of Mangold and Bisgaard renders claim 2 obvious.

137. Claim 2 depends from claim 1. Each of the hearing correction filters disclosed by Fichtl "comprises a collection of acoustic configuration settings configured to modulate the electrical signal to a level that is within a range between an uncompensated hearing level of the user and the level to compensate

for the hearing impairment of the user." Fichtl, Ex. 1003 at Abstract, 1:23-26, Figs. 2-4, 3:42-4:11 (describing continuous increase of the settings until the target value is reached), 5:11-34 (same), 5:42-6:18 (same).

138. Fichtl discloses an initial APP value (iPOV) and a target APP value (tPOV), and all of the other incremental APP values (rPOV), fall between the initial and target APP values, iPOV and tPOV, respectively. Fichtl, Ex. 1003 at Abstract, 1:23-26, Figs. 2-4, 3:42-4:11 (describing continuous increase of the settings until the target value is reached), 5:11-34 (same), 5:42-6:18 (same). The initial, incremental, and target APP values are used by the signal processor to modulate the audio signal received from the microphone. *Id.* at 3:27-28, 3:35-4:15, 4:25-67, Fig. 1. Therefore, the initial and incremental APP values each modulate the electrical signal to a level that is within a range between an uncompensated level and the level for optimal hearing compensation.

J. [Claim 3.1] "a transceiver coupled to the processor and configurable to communicate with a computing device through a communication channel during operation, the transceiver to receive a signal from the computing device and to provide the signal to the processor"

139. The combination of Fichtl, Mangold, and Bisgaard satisfies this limitation. Claim 3 depends from claim 1. The combination of Fichtl, Mangold, and Bisgaard renders claim 1 obvious as described above. When a user is treated

by an audiologist, specific audio processing parameters ("APPs") are chosen based on the user's hearing loss. Fichtl, Ex. 1003 at 3:27-41, Fig. 1.

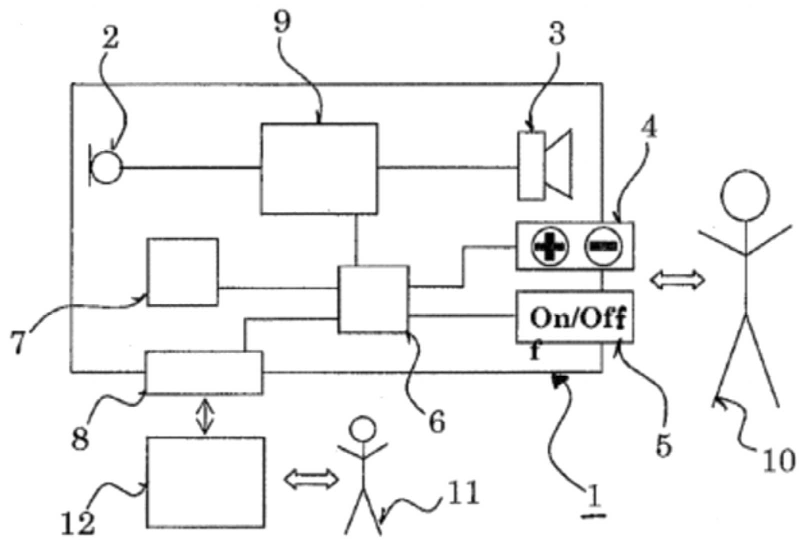


Fig. 1

Fichtl, Ex. 1003, Figure 1

140. The parameters are input by an audiologist (Fichtl, Ex. 1003 at Fig. 1, ref. no. 11) using a fitting interface (*id.* at Fig. 1, ref. no. 8). *Id*. at 13:36-37 ("a fitting interface (8) for adjusting said hearing device (1) to the needs of said hearing device user (10)"), 13:45-14:1 ("fitting interface (8) is adapted to write an initial power-on value (iPOV) and a value indicative of a target power-on value (tPOV) for said audio processing parameter (APP) to said non-volatile memory (7)"), 14:41-45 ("fitting said hearing device (1) . . . by a fitter programming an initial power-on value (iPOV) and the value indicative of said target power-on value (tPOV) for said audio processing parameter (APP) to said non-volatile memory (7)"), 3:21-30, Fig. 1. The fitting interface (8) is coupled to the controller

(6) which provides the APP values to signal processor (9) for processing signals. *Id*. The fitting interface (8) communicates with a computing device (12), which is operated by the audiologist (11). *Id*. at Fig. 1, Abstract, 3:35-48. The communication is represented by the bi-directional arrow between elements 8 and 12 in Fig. 1. *Id*. Thus, Fichtl discloses a transceiver (e.g., fitting interface (8)) coupled to the processor (e.g., controller (6)) and configurable to communicate with a computing device (e.g., device (12)) through a communication channel, the transceiver to receive a signal from the computing device and to provide a signal to the processor (e.g., bidirectional arrow).

141. Fichtl discloses that "[a]dditional devices such as a remote control may be considered to be part of the hearing device.". Fichtl, Ex. 1003 at 1:16-18. It would have been obvious to use Fichtl's transceiver, rather than a separate communications interface, to communicate with Fichtl's remote control in order to save space.

142. Mangold discloses that the hearing aid can have a "remote control unit" that includes a control unit, a transmitter, a manual program control, and, optionally, a memory. Mangold, Ex. 1007 at 1:59-2:2, 3:38-59, Fig. 3, 3:10-15, Fig. 5, 4:11-26. The hearing aid user can select a stored program using the manual program control 24, which is part of the remote control. *Id*. at 3:13-15, 3:38-43, Fig. 3, 4:11-16, Fig. 4. The selected program is stored in the memory (*Id*. at 3:10-

13, Fig. 1), which can be located in the hearing aid or in the remote control unit. *Id*. at 3:27-29, Fig. 2, 4:11-21, Fig. 5. The selected program can be identified by a program number. *Id*. at 3:49-59 (describing transmitting a "program number" identifying a selected program to the hearing aid when the memory is located in the hearing aid). Thus, Mangold discloses a computing device (remote control unit) that receives a selection from the hearing aid user during operation via the manual program control and, in response, sends a signal, such as an indicator identifying a selected program (if the memory is in the hearing aid) or the selected program itself (if the memory is in the remote control unit), to the hearing aid for receipt by a hearing aid processor to allow the hearing aid processor to implement the selected program.

143. It would have been obvious to modify Fichtl in view of Mangold to include the claimed transceiver as discussed below. In particular, it would have been obvious to modify Fichtl's remote control to include Mangold's manual program control 18, and Mangold's memory storing programs. When a hearing aid user turns on the hearing aid, the remote control would retrieve an APP power-on value from memory (Fichtl, Ex. 1003 at 3:26-27, 3:49-51) and transmit the APP power-on value to the hearing aid for use by the hearing aid processor to process audio signals. *Id*. at 3:27-28, 3:51-53, 4:6-7. Alternatively, Fichtl's memory 7 would be in the hearing aid, as taught by Mangold. In this case the hearing aid

would receive an indicator identifying the selected APP power-on value to implement from the remote control, as taught by Mangold, and in response the hearing aid processor would retrieve the identified APP power-on value from memory.

144. A POSA would have been motivated to implement Fichtl's remote control as taught by Mangold. Mangold notes several benefits regarding the use of remote controls with hearing aids, including the hearing aid "can be smaller, lighter in weight, and less visible" and more advanced components can be included in the remote control due to its larger size and greater battery power. Mangold, Ex. 1007 at 1:67-2:2, 4:21-26. Incorporating components, such as the memory, control unit, manual control switches, or other components, in the remote control was known in the art as of 2010. *Id.* The combination would have involved nothing more than combining prior art elements according to known methods, with no change in their respective functions, to yield predictable results.

K. [Claim 3.2] "wherein the processor applies the selected one of the sequence of incremental hearing correction filters in response to receiving the signal"

145. The combination of Fichtl and Mangold results in the second limitation of claim 3. As discussed above with respect to element 1.5.3, Fichtl discloses applying the selected second one of the sequence of incremental hearing correction filters in response to the hearing aid being turned on. It would have

been obvious in view of Mangold for Fichtl's on/off switch to be located on Fichtl's remote control to allow a hearing aid user to turn the hearing aid on and off using the remote control and for Fichtl's hearing aid to receive a signal in response to the hearing aid user turning on the hearing aid. Thus, it would have been obvious for the processor to apply the selected one of the sequence of incremental hearing correction filters in response to receiving the signal from the computing device, as recited in the second limitation of claim 3.

L. [Claim 4] "hearing aid of claim 3, wherein the signal includes the selected one of the sequence of incremental hearing correction filters"

146. The combination of Fichtl and Mangold results in a "hearing aid of claim 3, wherein the signal includes the selected one of the sequence of incremental hearing correction filters." As discussed above with respect to element 3.1, Fichtl as modified by Mangold discloses the hearing aid transceiver receiving from the remote control an APP power-on value for use by the hearing aid processor to process audio signals. As discussed above with respect to elements 1.5.2 and 1.5.3, each APP power-on value corresponds to a selected incremental hearing correction filter of the sequence of incremental hearing correction filters and is used by the hearing aid to adjust the hearing aid's processing of audio signals to a level of correction less than the target value tPOV. Fichtl, Ex. 1003 at Fig. 2, 4:41-51. Because this signal sent from the remote

control to the hearing aid comprises an adjustment that the hearing aid processor applies to its processing of signals to reduce the level of correction provided to a user, the signal includes the selected one of the sequence of incremental hearing correction filters.

M. [Claim 5.1] "hearing aid of claim 3, further comprising a memory to store the sequence of incremental hearing correction filters"

147. As explained above with respect to element 1.4, Fichtl alone or in view of Mangold discloses a hearing aid with a memory. Furthermore, Fichtl's memory stores the sequence of incremental hearing correction filters. Fichtl, Ex. 1003, at 3:30-32 ("There is a non-volatile memory 7 to store parameters . . ."), 3:66-4:4, 4:25-30, 4:33-36; *see also* Mangold, Ex. 1007 at 3:10-13, 2:67-3:2, Fig. 1.

N. [Claim 5.2] "wherein the signal includes an indicator identifying the selected one of the incremental hearing correction filters within the sequence"

148. The combination of Fichtl and Mangold render claim 5 obvious. As discussed above with respect to element 3.1, Fichtl as modified by Mangold discloses the signal including an indicator identifying the selected APP power-on value. As discussed above with respect to elements 1.5.2 and 1.5.3, each APP power-on value corresponds to a selected incremental hearing correction filter of the sequence of incremental hearing correction filters. Thus, Fichtl as modified by

Mangold discloses that the signal includes an indicator identifying the selected one of the incremental hearing correction filters within the sequence.

O. [Claim 5.3] "wherein, in response to receiving the signal, the processor retrieves the selected one of the incremental hearing correction filters from the memory and applies the selected one to the modulated electrical signals"

149. The combination of Fichtl and Mangold satisfies this claim element. As discussed above with respect to element 3.1, Fichtl as modified by Mangold discloses the hearing aid would receive an indicator identifying the selected APP power-on value to implement from the remote control, as taught by Mangold, and in response the hearing aid processor would retrieve the identified APP power-on value from memory. As discussed above with respect to elements 1.5.2 and 1.5.3, each APP power-on value corresponds to a selected incremental hearing correction filter of the sequence of incremental hearing correction filters and is applied by Fichtl's processor to the modulated electrical signals.

P. [Claim 16] "hearing aid of claim 1, further comprising instructions that, when executed by the processor, cause the processor to generate the sequence of incremental hearing correction filters based at least in part on a magnitude of a difference between a hearing aid profile and a hearing loss level associated with the user of the hearing aid, the sequence of incremental hearing correction filters including at least the first hearing correction filter and the second hearing correction filter"

150. The combination of Fichtl, Mangold, and Bisgaard renders claim 16 obvious.

151. As discussed above with respect to elements 1.5.2 and 1.5.3, Fichtl discloses an acclimatization program in which audio processing parameter (APP) settings are incrementally adjusted over time from an initial setting iPOV to a target setting tPOV, providing for at least first and second hearing correction filters. Fichtl, Ex. 1003 at 3:35-4:15, 4:25-67. Fichtl's hearing device "is initially fitted to a hearing loss of a hearing device user," including an initial power-on value iPOV and a target power-on value tPOV. *Id.* at 3:37-48. As discussed in Section VII.G. [element 1.5.2], the iPOV is selected to provide a smaller degree of compensation as compared to the tPOV, and the tPOV corresponds to a hearing aid profile. Thus, the iPOV and all incremental adjustments in-between the iPOV and the tPOV are generated to provide values between the hearing loss level associated with the user (determined during the initial fitting) and the hearing aid profile (tPOV). This sequence of incremental hearing correction filters is, accordingly, based at least in part on a magnitude of a difference between a hearing aid profile and a hearing loss level associated with the user of the hearing aid, and includes at least the first and second hearing correction filter, as claimed.

XII. CLAIM 18 IS OBVIOUS OVER FICHTL (EX. 1003) IN VIEW OF MANGOLD (EX. 1007), BISGAARD (EX. 1006), AND SACHA (EX. 1004).

A. [Claim 18.1] "hearing aid of claim 1, further comprising instructions that, when executed by the processor, cause the processor to: determine an amount of time during which the first hearing correction filter is applied"

152. Fichtl and Sacha combine to create a "hearing aid of claim 1, further comprising instructions that, when executed by the processor, cause the processor to: determine an amount of time during which the first hearing correction filter is applied." As explained above, Fichtl and Sacha each disclose a hearing aid with instructions for executing an acclimatization program.

153. Sacha discloses an acclimatization program and a time-based trigger for sequencing through intermediate, "sub-optimal" signal processing parameters. Sacha, Ex. 1004 at [0012]; *see also Id*. at Abstract, Fig. 2 (230), [0004] ("The device may be programmed to select a signal processing parameter set … in a defined sequence based upon elapsed operating time intervals as measured by a timer"), [0015]. Sacha uses a timer 230 that "operates when the device is powered on" to "record[] the time during which the device is powered up" to maintain a "running total of the operating time" that allows the hearing aid to switch to the next parameter set after a "specified operating time interval has elapsed." *Id*. at [0015]. Thus, Sacha discloses determining an amount of time during which a hearing correction filter of the sequence is applied.

154. It would have been obvious to modify Fichtl in view of Sacha to include Sacha's timer to allow the second hearing correction filter to be triggered after applying the first hearing correction filter for a predetermined amount of time, as taught by Sacha, instead of or in addition to in response to the hearing aid user

turning the hearing aid off then on. Fichtl itself contemplates applying the next APP value before the next power-on event. Fichtl, Ex. 1003 at 4:20-24 ("it does not matter if the change due to the acclimatization algorithm is already applied during the current usage period, or . . . not until the hearing device 1 is switched off and on again"). A POSA would have recognized that power-on events and operating time are each common, alternative mechanisms for triggering the next incremental correction during an acclimatization process. *See* Sacha, Ex. 1004 at [0004] ("The device may be programmed to select a signal processing parameter set for specifying to the signal processing circuit from a group of such parameter sets in a defined sequence ***based upon elapsed operating time intervals as measured by a timer or upon a specified number of detected power events representing the device being turned on***."), [0013]-[0015]. Such a modification prevents the acclimatization process from occurring too quickly, if the hearing aid user powers on and off frequently, or too slowly, if the hearing aid user powers on and off infrequently. *See* Fichtl, Ex. 1003 at 2:13-18, 1:57-2:10. Triggering subsequent incremental corrections based on time intervals also allows control over which corrections are applied for longer time intervals, such as when the hearing aid user first starts wearing the hearing aid, and which are applied for shorter time intervals, such as when the corrections are closer to the target hearing compensation. *See*, Sacha, Ex. 1004 at [0015] (disclosing different time intervals for the sequence of incremental corrections).

B. [Claim 18.2] "apply the second hearing correction filter when the amount of time exceeds a pre-determined threshold"

155. The combination of Fichtl and Sacha satisfies this claim element. As explained above with respect to element 18.1, Fichtl in view of Sacha discloses applying the second hearing correction filter when the amount of time exceeds a pre-determined threshold.

XIII. CLAIMS 6-9 AND 17 ARE OBVIOUS OVER FICHTL (EX 1003) IN VIEW OF SACHA (EX 1004), MANGOLD (EX. 1007), AND DE19542961 (EX. 1009)

156. As provided in my detailed analysis below, it is my opinion that claims 6-9 and 17 of the '999 patent are rendered obvious by Fichtl in view of Sacha, Mangold, and DE19542961.

A. [Claim 6.1] "A non-transitory computer-readable device comprising instructions that, when executed by a processor, cause the processor to…"

157. As explained above with respect to element 1.4, Fichtl alone or in view of Mangold discloses a memory to store instructions. Because a memory storing instructions is a non-transitory computer-readable device comprising instructions executable by a processor, Fichtl alone or in view of Mangold discloses this limitation.

C. [Claim 6.2] "select a hearing aid profile from a plurality of hearing aid profiles, the selected hearing aid profile configured to modulate an audio signal to a level to compensate for a hearing impairment of a user"

158. As explained above with respect to element 1.5.1, the combination of Fichtl and Mangold results in a hearing aid with a processor that selects a hearing aid profile from a plurality of hearing aid profiles, the selected hearing aid profile configured to modulate an audio signal to a level to compensate for a hearing impairment of a user.

D. [Claim 6.3] "apply a first hearing correction filter to the selected hearing aid profile to reduce the amplitude of the modulated audio signal produced by the selected hearing aid profile to a first level that is less than the level to compensate for the hearing impairment of the user"

159. As explained with respect to elements 1.5.1 and 1.5.2, Fichtl discloses this limitation.

E. [Claim 6.4] "determine an amount of time during which the first hearing correction filter is applied"

160. As explained above with respect to element 18.1, Fichtl in view of Sacha discloses this limitation.

F. [Claim 6.5.1] "selectively apply a second hearing correction filter to the selected hearing aid profile to reduce the amplitude of the modulated audio signal produced by the selected hearing aid profile to a second level that is greater than the first level and less than the level to compensate for the hearing impairment of the user when the amount of time exceeds a pre-determined threshold"

161. Fichtl in view of Sacha disclose this limitation. As explained above with respect to elements 1.5.3 and 18.2, Fichtl in view of Sacha discloses applying a second hearing correction filter, or APP, to the selected hearing aid profile to reduce the amplitude of the modulated audio signal produced by the selected hearing aid profile to a second level that is greater than the first level and less than the level to compensate for the hearing impairment of the user in response to a trigger is when a pre-determined time threshold is exceeded.

G. [Claim 6.5.2] "the pre-determined threshold is programmable by the user"

162. The combination of Fichtl, Sacha, and DE961 renders this limitation obvious. DE961 (Ex. 1009) discloses a hearing aid with an acclimatization feature. DE961, Ex. 1009 at 1-3.

163. DE961 teaches that the pre-determined threshold is programmable directly by the user. DE961 discloses a control unit with a timer that determines the time for each interim level (that is, the "pre-determined threshold"). DE961, Ex. 1009 at 3 ("Appropriately, the adjustment of the operating parameter or operating parameters occurs in uniform steps in order of magnitude and/or in time

intervals. In this case, the control unit comprises a timing element or timer for adjustment of a time constant”). The hearing aid user can adjust the timing by means of an “adjustment program.” *Id.* at 3 (“the patient himself or herself can choose a longer time constant”); 4-5 (“It is possible with the timer 15 for the patient to preferably select alternatively a plurality of time constants. … the patient can select a slower time constant”); 5 (the parameters can be adjusted “on an individual and patient-related basis by means of an adjustment program”). Accordingly, the pre-determined threshold of time is programmable by the user.

H. A POSA Would Have Combined Fichtl, Sacha, and DE961

164. A POSA would have recognized that Fichtl, Sacha, Mangold, and DE961 all disclose hearing aids with acclimatization programs. Fichtl, Ex. 1003 at Abstract; Sacha, Ex. 1004 at Abstract, [0003], [0004]; DE961, Ex. 1009, at 1, 2-3.

165. Mangold discloses a hearing aid system, which can include an auditory prosthesis for the user’s ear and a remote control, that contains programmable memory for storing a plurality of “programs,” which include a collection of parameters and/or coefficients in a hearing aid program for processing audio signals. Mangold, Ex. 1007 at 1:9-22, 2:28-33, 4:47-58, 1:23-29. As described above with respect to Claims 1-5 and 16, there are many reasons a POSA would have combined Fichtl and Mangold. In addition, a POSA would have been motivated to modify Fichtl’s hearing aid according to the disclosure of Sacha, as

explained above with respect to element 18.1. DE961's disclosure that a user controls the length of time for which each hearing correction filter for various parameters is applied could and would be incorporated to control the acclimatization program of Fichtl's hearing aid as modified by Sacha and Mangold. *Id.* As noted in Fichtl, one disadvantage of prior hearing devices was the hearing device's inability to take user preferences into account, including the inability to determine if the user prefers a faster or a slower increase of the intensity of the hearing device. Fichtl, Ex. 1003 at 1:41-47. Allowing for some user control of the acclimatization program, such as how quickly incremental hearing correction filters are applied, ensures that the program is working for the hearing aid user and is not adjusting too slowly or too quickly, which can affect whether the user actually uses the hearing aid. *See*, *e.g.*, Fichtl, Ex. 1003 at 2:52-60 (describing allowing user influence on an acclimatization process as beneficial); DE961, Ex. 1009 at 3 ("the ***patient himself or herself can choose a longer time constant***, that is, a longer time interval, ***in the event that the adjustment . . . is too fast***"), 5 ("For the case when an adjustment of the operating parameters of the starting situation to the target situation occurs ***too rapidly for the patient***, ***the patient can select a slower time constant*** of the circuit."). Both the hearing device disclosed in Fichtl and in DE961 include options for obtaining input from the hearing aid user, for example, through the volume control or through a remote control or other external input unit. *See*, *e.g.*, Fichtl, Ex. 1003 at 3:25-26, 1:16-18, Fig. 1; DE961, Ex. 1009 at 4-5, Figs. 1 and 2. Thus, the hearing device taught by Fichtl could and would be modified to include the user control disclosed by DE961.

166. The combination of Fichtl's hearing aid system with the disclosures of Sacha, Mangold, and DE961 would have involved nothing more than combining prior art elements, as described above, according to known methods, with no change in their respective functions, to yield predictable results.

I. [Claim 7] "computer-readable device of claim 6, wherein the pre-determined threshold is configurable by the user"

167. The combination of Fichtl, Sacha, Mangold, and DE961 renders claim 7 obvious.

168. As discussed above with respect to element 6.5.2, DE961 discloses that the patient (hearing aid user) can select the timing (pre-determined threshold); therefore, DE961 discloses that the "computer-readable device of claim 6, wherein the pre-determined threshold is configurable by the user."

J. [Claim 8] "computer-readable device of claim 6, further comprising instructions that, when executed by the processor, cause the processor to receive the first hearing correction filter and the second hearing correction filter from a transceiver configured to communicatively couple to a computing device during operation"

169. The combination of Fichtl, Sacha, Mangold, and DE961 renders claim 8 obvious.

170. Fichtl in view of Mangold discloses this limitation. As described above with respect to elements 1.4 – 1.5.3, Fichtl in view of Mangold discloses a memory to store instructions to be executed by the processor to cause the processor to receive the first and second hearing correction filters. As described above with respect to elements 3.1 and 3.2 and claim 4, Fichtl in view of Mangold discloses a transceiver configured to communicatively couple to a computing device during operation including receiving from the computing device hearing correction filters. The hearing correction filters are executed by controller 6. Fichtl, Ex. 1003 at 3:27-30, 4:25-26. Thus, the hearing correction filters, to be applied, would be received by controller 6 of Fichtl (the claimed processor), which is in communication with fitting interface 8 [transceiver] of Fichtl (Fichtl, Ex. 1003 at Fig. 1).

K. [Claim 9] "non-transitory computer-readable device of claim 6, further comprising instructions that, when executed by the processor, cause the processor to dynamically generate the first hearing correction filter and the second hearing correction filter based on at least one of the hearing impairment of the user and a hearing aid profile including a collection of acoustic configuration settings for producing the modulated output signal at the corrected hearing level"

171. The combination of Fichtl, Sacha, Mangold, and DE961 renders claim 9 obvious.

172. Fichtl discloses this limitation. The acclimatization algorithm disclosed by Fichtl includes the creation of an intermediate value X which becomes the next, replacement power-on APP, the rPOV corresponding to the next hearing correction filter. Fichtl, Ex. 1003 at 3:35-4:15. The APPs are dynamically generated based off the "target power-on value" tPOV which forms part of the hearing aid profile. *Id.* at 1:14-15, 1:36-38, 2:40-49 (acclimatization is "designed to approximate said power-on value (POV) in the long term"). For example, Fichtl discloses the initial power-on APP value (iPOV) as 10dB lower than the target power-on value (tPOV). *Id.* at 3:37-4:15. Each replacement power-on value (rPOV) corresponding to a subsequent hearing correction filter is dynamically generated using recursive functions based on the preceding power-on value. *See, e.g.*, *Id.* at 5:20-34 (exemplary recursive function for dynamically generating rPOV), 5:53-6:19 (exemplary recursive functions for dynamically generating rPOV).

L. [Claim 17] "non-transitory computer-readable device of claim 6, further comprising instructions that, when executed by the processor, cause the processor to generate the sequence of incremental hearing correction filters based at least in part on a magnitude of a difference between a hearing aid profile and a hearing loss level associated with the user of the hearing aid, the sequence of incremental hearing correction filters including at least the first hearing correction filter and the second hearing correction filter"

173. The combination of Fichtl, Sacha, Mangold, and DE961 renders this claim obvious. As discussed above with respect to claim 16, Fichtl discloses this limitation.

XIV. CLAIM 19 IS OBVIOUS OVER FICHTL (EX. 1003) IN VIEW OF SACHA (EX. 1004), MANGOLD (EX. 1007), BISGAARD (EX. 1006), AND DE19542961 (EX. 1009)

174. As provided in my detailed analysis below, it is my opinion that claim 19 of the '999 patent are rendered obvious by Fichtl in view of Sacha, Mangold, Bisgaard, and DE961.

M. [Claim 19] "hearing aid of claim 18, wherein the pre-determined threshold is adjustable by the user"

175. The combination of Fichtl, Sacha, Mangold, Bisgaard, and DE961 renders claim 19 obvious.

176. As discussed above, the combination of Fichtl, Mangold, Bisgaard, and Sacha renders claim 18 obvious. As further discussed above with respect to claims 6 and 7, DE961 discloses that the user can select the timing constant. A POSA would have been motivated to modify the combination of Fichtl, Mangold, Bisgaard, and Sacha so that the threshold is adjustable by the user, as taught by DE961 for the same reasons as discussed above with respect to claims 6 and 7.

177. The combination of Fichtl's hearing aid system with the disclosures of Mangold, Bisgaard, Sacha, and DE961 would have involved nothing more than

combining prior art elements, as described above, according to known methods, with no change in their respective functions, to yield predictable results.

XV. CLAIMS 10, 13, 14 AND 20 ARE OBVIOUS OVER FICHTL (EX. 1003) IN VIEW OF MANGOLD (EX. 1007)

178. As provided in my detailed analysis below, it is my opinion that claims 10, 13, 14, and 20 of the '999 patent are rendered obvious by Fichtl in view of Mangold.

A. [Claim 10 - preamble] "A computing device comprising"

179. Fichtl in view of Mangold discloses this limitation. Fichtl discloses a hearing aid that can be used with a remote control. Fichtl, Ex. 1003 at 1:14-18 ("Additional devices such as a remote control may be considered to be part of the hearing device."). Although Fichtl does not provide details on the remote control, Mangold does. Mangold is directed to "auditory prostheses including hearing aids." Mangold, Ex. 1007 at 1:9-15. Mangold discloses a hearing aid that has a "remote control unit" that includes a control unit, a transmitter, a manual program control, and a memory. *Id.* at 1:59-2:2, 3:38-49, Fig. 3, 3:10-15, Fig. 5, 4:11-26. Mangold's remote control constitutes a "computing device."

180. It would have been obvious to implement Fichtl's remote control as a "computing device" as taught by Mangold. For example, it would have been obvious to implement Fichtl's hearing device such that Fichtl's user controls, controller to determine audio processing parameters (APPs), and memory to store

the APPs are implemented in Fichtl's remote control, as Mangold discloses implementing similar or analogous components, used for a similar purpose, in a remote control. In particular, Mangold discloses a remote control including user controls (i.e., "manual program control") for selecting a program comprising parameter settings for processing audio signals. Mangold, Ex. 1007 at 2:28-33, 3:13-15, 3:40-41, Fig. 3, 4:11-16, Fig. 5. Similarly, Fichtl discloses user controls (i.e., APP control 4, on/off switch 5) for selecting or changing APP values used to process audio signals. Fichtl, Ex. 1003 at 3:25-30, 3:44-47, 3:58-60, 3:62-64, 4:5-7, 4:33-40, Fig. 1. Mangold further discloses a remote control including a control unit or logic (i.e., logic block 26) for selecting or calculating a program. Mangold, Ex. 1007 at 1:59-66, 3:40-43, 3:49-66, Fig. 3, 4:11-16, Fig. 5. Similarly, Fichtl discloses controller 6 for setting APP values. Fichtl, Ex. 1003 at 3:27-30, 3:32-34, 4:25-53, Fig. 1. Mangold further discloses a remote control including a memory storing programs. Mangold, Ex. 1007 at Abstract, 3:10-13, 4:11-21, Fig. 5. Similarly, Fichtl discloses a memory for storing APP values. Fichtl, Ex. 1003 at 3:30-32, 3:50-53, 3:67-4:4, 4:13-15, 4:28-30, 4:34-36, Fig. 1. Together, the user controls, controller, and memory of both Fichtl and Mangold allow a hearing aid user to control parameter values or settings used to process audio signals for the hearing aid user. Thus, implementing Fichtl's remote control, as taught by

Mangold, would allow a hearing aid user to control APP values using the remote control.

181. A POSA would have been motivated to incorporate Fichtl's user controls, controller, and memory in Fichtl's remote control, as taught by Mangold for the following reasons.

182. First, both Fichtl and Mangold are in the same field—hearing aid devices, address similar problems—insuring the comfort of hearing aid users, and provide similar solutions—adjusting parameters used by the hearing aid to process audio signals in response to user input. Fichtl, Ex. 1003 at Abstract; Mangold, Ex. 1007 at Abstract. A POSA would thus have considered the disclosure of Fichtl and Mangold together. For example, a POSA would have been motivated to combine the hearing aid and remote control of Mangold with an acclimatization system to provide the hearing aid user with a gradual transition from no hearing aid to full compensation for the user's loss of hearing. Bisgaard, Ex. 1006 at 1:5-15 (describing using hearing aid of Mangold), 2:48-56 (describing using the hearing aid with a "habituation system" to provide "a gradual transition from no hearing aid to full compensation for the user's loss of hearing").

183. Second, there are several benefits to incorporating hearing aid components such as user controls, a processor, and a memory in a remote control external to the hearing aid. Mangold notes that the hearing aid "can be smaller,

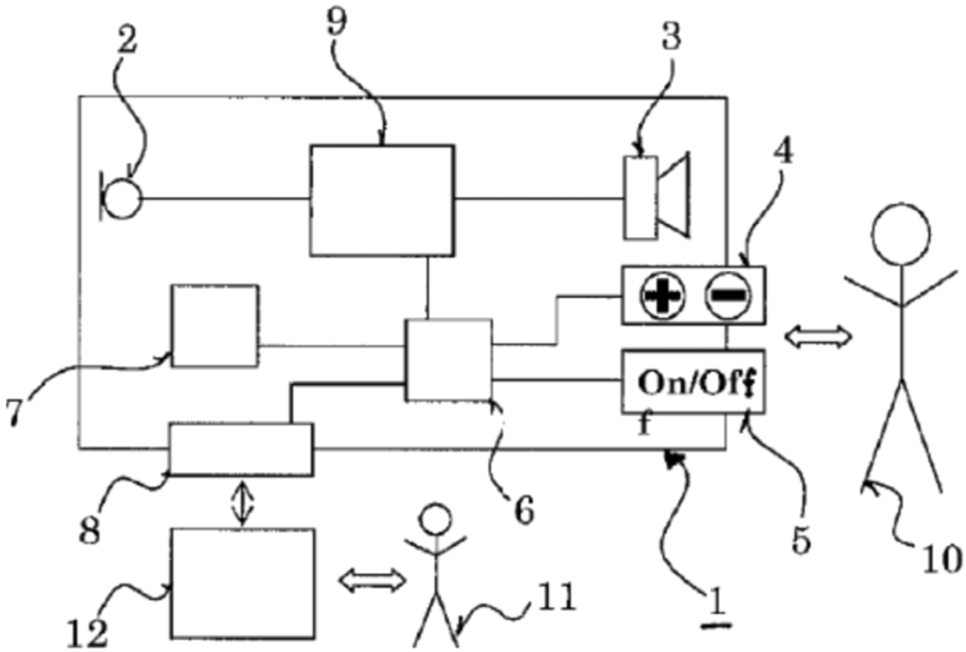
lighter in weight, and less visible" and that more advanced components can be included in the remote control due to its larger size and greater battery power. Mangold, Ex. 1007 at 1:67-2:2, 4:21-26. Furthermore, such a remote control would allow the hearing aid user to fit the hearing aid himself, reducing the amount of help needed from an audiologist. Roeck, Ex. 1010 at 1:58-64, 3:2-3, 3:4-10, 3:34-38, 7:1-20. Thus, a POSA would have been motivated to implement Fichtl's remote control as taught by Mangold by incorporating components of the hearing device into the remote control.

184. Finally, moving components, such as the memory, control unit, manual control switches, or other components, from a location physically in the hearing aid to the remote control was known in the art as of 2010. Mangold, Ex. 1007 at 4:21-26; *see also* Roeck, Ex. 1010 at 3:4-10, 3:34-38, 4:4-9. The combination would have involved nothing more than combining prior art elements according to known methods, with no change in their respective functions, to yield predictable results.

B. [Claim 10.1] "a transceiver configurable to communicate with a hearing aid through a communication channel"

185. Fichtl alone or in view of Mangold discloses this limitation. Fichtl discloses a transceiver in a hearing aid configurable to communicate with a transceiver in a computing device. In particular, Fichtl discloses an audiologist programs into the hearing aid audio processing parameters (APPs), such as

"volume," "treble or noise canceling," chosen for the user's hearing loss. Fichtl, Ex. 1003 at 3:27-48, Abstract ("An initial power-on value (iPOV) and a target power-on value (tPOV), which is to be reached at the end (H) of the acclimatization phase, may be programmed by an audiologist."). The parameters are input by an audiologist (*id.* at Fig. 1, ref. no. 11) using a fitting interface (*id.* at Fig. 1, ref. no. 8). *Id*. at 13:36-37 ("a fitting interface (8) for adjusting said hearing device (1) to the needs of said hearing device user (10)"), 13:45-14:1 ("fitting interface (8) is adapted to write an initial power-on value (iPOV) and a value indicative of a target power-on value (tPOV) for said audio processing parameter (APP) to said non-volatile memory (7)"), 14:41-45 ("fitting said hearing device (1) . . . by a fitter programming an initial power-on value (iPOV) and the value indicative of said target power-on value (tPOV) for said audio processing parameter (APP) to said non-volatile memory (7)"), 3:21-30, Fig. 1. The fitting interface (8) communicates with a computing device (12), which is operated by the audiologist (11). *Id*. at Fig. 1, Abstract, 3:35-48. The communication is represented by the bi-directional arrow in Fig. 1. *Id*. Thus, Fichtl discloses a transceiver (e.g., fitting interface (8)) in a hearing aid configurable to communicate with a transceiver in a computing device (e.g., computing device (12)).



Fichtl, Ex. 1003, Fig. 1

186. Fichtl also discloses that the hearing device includes a remote control. Fichtl, Ex. 1003 at 1:14-18. It would have been obvious, in order to save space, to use Fichtl's transceiver, rather than a separate communications interface, to communicate with Fichtl's remote control. Fichtl's remote control necessarily includes a mechanism to communicate with the hearing aid. A transceiver is a well-known mechanism for communicating between electronic devices such as hearing aids and remotes, and allows the remote control not only to send commands and data, but also to receive confirmations and other data from the hearing aid.

187. Furthermore, the combination of Fichtl and Mangold renders obvious this limitation. As discussed above with respect to element 10 – preamble, a POSA would have been motivated to implement Fichtl's remote control to include Fichtl's user controls 4 and 5, controller 6 to determine audio processing parameters (APPs), and memory 7 to store the APPs. Fichtl's remote control would thus include at least a transmitter for sending via a communication channel, for example, APP values determined by controller 6 to the hearing aid to be used by the hearing aid's processor 9 to process audio signals. Fichtl, Ex. 1003 at 3:27-30, Fig. 1. That Fichtl's remote control would include such a transmitter for sending information used by the hearing aid to process audio signals is confirmed by Mangold, which also discloses such a transmitter. Mangold, Ex. 1007 at 1:61-66, 3:40-43, 3:46-49, 3:57-66, Fig. 3. Fichtl's remote control would also include a receiver for receiving via the communication channel, for example, information about the current sound environment based on signals detected by Fichtl's microphone located in the hearing aid. Fichtl, Ex. 1003 at 3:23-25, Fig. 1. Fichtl's controller 6 would be modified to determine APP values based on such information, as taught by Mangold. Mangold, Ex. 1007 at 1:45-49, 3:49-66. A POSA would have been motivated to modify Fichtl's remote control to include such a receiver to adapt the APP values determined by the controller to better compensate for the hearing aid user's hearing loss according to the user's current

sound environment. Thus, Fichtl's remote control would include a transceiver configurable to communicate with a hearing aid through a communication channel.

C. [Claim 10.2] "a processor coupled to the transceiver"

188. Fichtl in view of Mangold discloses this limitation. As discussed above with respect to element 10 – preamble, a POSA would have been motivated to implement Fichtl's remote control to include Fichtl's user controls 4 and 5, controller 6 to determine audio processing parameters (APPs), and memory 7 to store the APPs. Fichtl's controller can execute software corresponding to an acclimatization algorithm and thus comprises a processor. Fichtl, Ex. 1003 at 3:32-34, 4:25-26. Furthermore, Fichtl's signal processor 9 in the hearing aid uses APPs determined by controller 6 to process audio signals to be presented to the hearing aid user. *Id*. at 3:23-25, 3:27-30, Fig. 1. Thus, Fichtl's controller, as modified in view of Mangold to be incorporated in the remote control, would be coupled to the remote control's transceiver to allow APPs to be communicated to the hearing aid.

D. [Claim 10.3] "a memory coupled to the processor and configured to store instructions…"

189. Fichtl in view of Mangold discloses this limitation. As discussed above with respect to element 10 – preamble, a POSA would have been motivated to implement Fichtl's remote control to include Fichtl's user controls 4 and 5, controller 6 to determine audio processing parameters (APPs), and memory 7 to

store the APPs. Fichtl's controller can execute software corresponding to an acclimatization algorithm. Fichtl, Ex. 1003 at 3:32-34, 4:25-26. A POSA would have understood that memory 7 (or a separate memory) would have been configured to store instructions of the acclimatization program and coupled to the controller.

E. [Claim 10.4.1] "…instructions that, when executed by the processor, cause the processor to: generate a sequence of incremental hearing correction filters based at least in part on a magnitude of a difference between a hearing aid profile and a hearing loss level associated with a user of the hearing aid, the sequence of incremental hearing correction filters including at least a first hearing correction filter and a second hearing correction filter"

190. Fichtl in view of Mangold discloses this limitation. As discussed above with respect to element 10 – preamble, a POSA would have been motivated to implement Fichtl's remote control to include Fichtl's user controls 4 and 5, controller 6 to determine audio processing parameters (APPs), and memory 7 to store the APPs.

191. Fichtl's controller is programmed to execute an acclimatization algorithm where the amount of compensation for the user's hearing loss increases over time. Fichtl, Ex. 1003 at Abstract ("The intensity of the hearing device is increased in the long term"), 3:32-34 ("controller 6 is adapted to execute an acclimatization algorithm . . ."), 4:25-26 ("acclimatization process is controlled by software being executed on the controller 6"). As represented by the curve marked

"X" plotted on the graph depicted in Fig. 2, the acclimatization algorithm executed by controller 6 increases the value of an APP over time. *Id*. at Fig. 2, 3:35-36 ("FIG. 2 shows how an audio processing parameter APP is changed over time in a hearing device 1"), 3:42-4:15, 4:25-67. In particular, an intermediate value X is increased slowly while the hearing aid is on and held constant, stored in memory, while the hearing aid is off, such that each time the hearing aid is turned on the APP is set to the last value for X as stored in memory. *Id*. at Fig. 2, 3:55-57, 3:66-4:7, 4:31-36, 4:41-53.

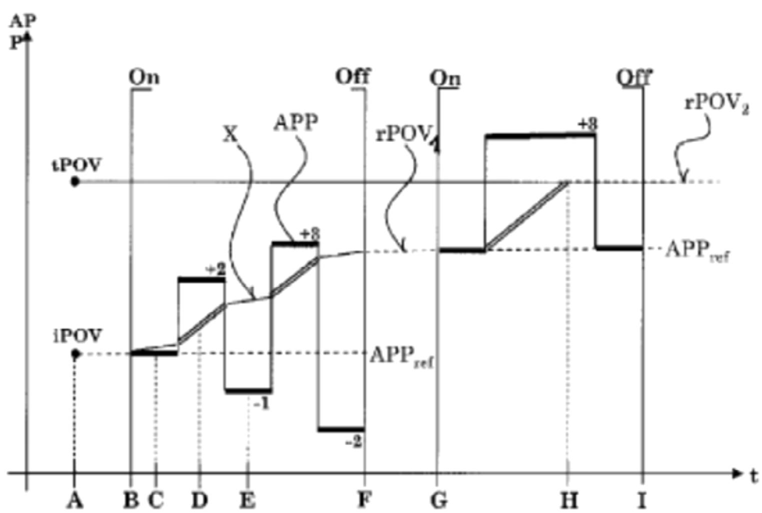


Fig. 2

192. As shown in Fig. 2, the APP starts at an initial power-on value (iPOV) selected to provide a smaller degree of compensation as compared to the target power-on value (tPOV) for the APP, which is the value for the APP corresponding to the selected hearing aid profile that compensates for the user's hearing

impairment, and the compensation increases over time to a replacement power-on-value (rPOV) each time the hearing aid is turned on until it reaches tPOV. Fichtl, Ex. 1003 at Fig. 2, 3:42-48 ("At time 'A,' a fitter programs an initial power-on value iPOV for the audio processing parameter as well as a target power-on value tPOV…The target power-on value tPOV is, for example, 10 dB higher than the initial power-on value iPOV), Abstract ("An initial power-on value (iPOV) and a target power-on value (tPOV), which is to be reached at the end (H) of the acclimatization phase, may be programmed by an audiologist."), 3:49-4:24, 4:25-67. Processor 9 uses APP values provided by controller 6, including the reduced values iPOV and rPOV relative to tPOV generated by the acclimatization algorithm, to process sounds for the hearing device user. *Id*. at Fig. 2, 3:23-34. Thus, Fichtl's acclimatization algorithm corresponds to adjustments applied by controller 6 to the collection of APPs of processor 9 to reduce the level of correction provided to the hearing device user by application of the hearing aid profile. In other words, Fichtl's acclimatization algorithm as executed by controller 6 generates a sequence of "hearing correction filters" including at least a first hearing correction filter and a second hearing correction filter.

193. The APP adjusted by the acclimatization algorithm may correspond to, for example, volume or treble. Fichtl, Ex. 1003 at 3:42-47. A volume APP corresponds to the loudness, or amplitude, of the output signal. *Id*. at 3:25-26

("The magnitude of the amplification can be controlled by a volume control 4."), 3:44-48 ("The audio processing parameter APP is typically volume . . . . The target power-on value tPOV is, for example, 10 dB higher than the initial power-on value iPOV."), 6:42-48 ("The adjustments in the first adjustment direction are implemented by applying a faster learning speed . . . . If the audio processing parameter APP is volume, the first adjustment direction is louder . . ."). A treble APP corresponds to the loudness, or amplitude, specifically of the higher frequencies. By applying an intermediate value that is lower than a target value, tPOV, for the volume or treble APP, Fichtl provides a modulated output signal having a level that is within a range between an uncompensated output level and the desired output level. Thus, Fichtl's hearing correction filters are "incremental hearing correction filters" and are created as in the '999 patent. '999 patent, Ex. 1001 at 3:32-41.

194. As discussed above, Fichtl discloses an acclimatization program in which audio processing parameter (APP) settings are incrementally adjusted over time from an initial setting iPOV to a target setting tPOV, providing for at least first and second hearing correction filters. Fichtl, Ex. 1003 at 3:35-4:15, 4:25-67. Fichtl's hearing device "is initially fitted to a hearing loss of a hearing device user," including an initial power-on value iPOV and a target power-on value tPOV. *Id.* at 3:37-48. The iPOV is selected to provide a smaller degree of compensation

as compared to the tPOV, and the tPOV corresponds to a hearing aid profile. Thus, the iPOV and all incremental adjustments in-between the iPOV and the tPOV are generated to provide values between the hearing loss level associated with the user (determined during the initial fitting) and the hearing aid profile (tPOV). This sequence of incremental hearing correction filters is, accordingly, based at least in part on a magnitude of a difference between a hearing aid profile and a hearing loss level associated with the user of the hearing aid, and includes at least the first and second hearing correction filter, as claimed.

F. [Claim 10.4.2] "provide a first signal related to the first hearing correction filter of the sequence of incremental hearing correction filters to the hearing aid through the communication channel"

195. Fichtl in view of Mangold discloses this limitation. As discussed above with respect to element 10.4.1, Fichtl's controller executes an acclimatization algorithm that generates power-on values for an audio processing parameter (APP) to be applied by a processor of the hearing aid, the sequence of power-on values corresponding to a sequence of incremental hearing correction filters. Fichtl's acclimatization algorithm is executed by Fichtl's controller 6 which would, as discussed above with respect to element 10 – preamble, be incorporated into Fichtl's remote control and, as discussed above with respect to element 10.2, communicate with Fichtl's hearing aid through a communication channel to provide audio processing parameter (APP) power-on values (rPOV) to a

processor of Fichtl's hearing aid. Fichtl, Ex. 1003 at 3:23-25, 3:27-30, Fig. 1. Because the APP power-on values provided by Fichtl's remote control to the hearing aid through a communication channel are related to corresponding incremental hearing correction filters, Fichtl in view of Mangold discloses providing a first signal related to the first hearing correction filter of the sequence of incremental hearing correction filters to the hearing aid through the communication channel.

G. [Claim 10.4.3] "provide a second signal related to a second hearing correction filter of the sequence of incremental hearing correction filters to the hearing aid in response to receiving a selection of the second hearing correction filter from a user of the hearing aid"

196. Fichtl in view of Mangold discloses this limitation. For the same reasons as discussed above with respect to element 10.4.2 with regards to providing a first signal related to a first hearing correction filter of the sequence to the hearing aid, Fichtl in view of Mangold discloses providing a second signal related to a second hearing correction filter of the sequence of incremental hearing correction filters to the hearing aid.

197. Furthermore, for the reasons discussed above with respect to element 10 – preamble, Fichtl's remote control, as implemented in view of Mangold, includes user controls including on/off switch 4 for allowing the user to turn the hearing aid on and off. *See* Fichtl, Ex. 1003 at 3:26-27, Fig. 1, 13:38-39. Fichtl

discloses application of the replacement power-on value, corresponding to the subsequent (e.g., second) hearing correction filter, is triggered in response to the hearing aid being turned on by the hearing aid user. Fichtl, Ex. 1003 at Fig. 2, 3:66-4:7, 4:31-36, 4:41-53. Thus, when the hearing aid user turns on the hearing aid via the on/off switch, the second hearing correction filter is selected. Thus, Fichtl in view of Mangold discloses providing a second signal related to a second hearing correction filter to the hearing aid in response to receiving a selection of the second hearing correction filter from a user of the hearing aid.

H. A POSA Would Have Combined Fichtl and Mangold

198. The combination of Fichtl and Mangold as provided for claim 10 and discussed below for dependent claims would have involved nothing more than combining prior art elements according to known methods with no change in their respective functions, to yield predictable results, such as the use of an external unit computing device with a processor coupled to the transceiver in order to communicate with the hearing aid. In addition, the prior art references are based on adjustment of a user's hearing aid so that it may be used comfortably. The addition of user input, either directly or indirectly, to the hearing aid of Fichtl would simply combine known prior art elements to yield predictable results. These predictable results would have also solved the recognized problem of maximizing a

hearing aid user's comfort by allowing the user to more easily adjust the hearing aid, possibly even without the help of an audiologist.

I. [Claim 13] "computing device of claim 10, wherein the first signal and the second signal comprise triggers to initiate an adjustment to a currently selected incremental hearing correction filter executing on the hearing aid"

199. As discussed above with respect to elements 10.4.2 and 10.4.3, Fichtl's remote control, as modified in view of Mangold, sends signals to the hearing aid that comprise triggers to initiate application of the subsequent incremental hearing correction filter, or initial hearing correction filter if at the start of acclimatization, by the hearing aid. Thus, these signals, including the first and second signals related to the first and second hearing correction filters, respectively, comprise triggers to initiate an adjustment to a currently selected incremental hearing correction filter executing on the hearing aid, as recited in this claim.

J. [Claim 14] "computing device of claim 10, wherein the first signal and the second signal include the first hearing correction filter and the second hearing correction filter"

200. As described above with respect to elements 10.4.2 and 10.4.3, Fichtl's remote control, as modified in view of Mangold, sends signals to the hearing aid that comprise APP power-on values (rPOV) each corresponding to a hearing correction filter of the sequence of hearing correction filters, including the first and second such filters. These APP power-on values are used by the hearing

aid to adjust the hearing aid's processing of audio signals to a level of correction less than the target value tPOV. Fichtl, Ex. 1003 at Fig. 2, 4:41-51. Because these signals sent from the remote control to the hearing aid comprise adjustments that the hearing aid processor applies to its processing of signals to reduce the level of correction provided to a user, as described above with respect to elements 10.4.2 and 10.4.3, the first and second of these signals include the corresponding first and second hearing correction filters, as recited in claim 14.

K. [Claim 20] "computer-readable device of claim 10, further comprising instructions that, when executed by the processor, cause the processor to receive: a selection of a hearing aid profile; and provide the hearing aid profile to the hearing aid"

201. Fichtl in view of Mangold discloses this claim. Fichtl discloses fitting a hearing aid to a user. Fichtl, Ex. 1003 at 3:37-39 ("The hearing device 1 is initially fitted to a hearing loss of a hearing device user 10 . . .").

202. Fichtl discloses that during the audiological fitting, various APPs, such as "volume," "treble or noise canceling" are chosen for the user's hearing loss. Fichtl, Ex. 1003 at 3:27-48, Abstract ("An initial power-on value (iPOV) and a target power-on value (tPOV), which is to be reached at the end (H) of the acclimatization phase, may be programmed by an audiologist."), Fig. 1 (depicting fitting interface (8) that communicates with computing device (12) operated by audiologist (11)), 13:36-37 ("a fitting interface (8) for adjusting said hearing device (1) to the needs of said hearing device user (10)"), 13:45-14:1 ("fitting

interface (8) is adapted to write an initial power-on value (iPOV) and a value indicative of a target power-on value (tPOV) for said audio processing parameter (APP) to said non-volatile memory (7)"). These APPs constitute a collection of acoustic configuration settings for a hearing aid which are used by Fichtl's signal processor 9 to shape acoustic signals to correct for a user's hearing loss and thus correspond to the claimed "hearing aid profile." *Id*. at 3:23-28 ("Sounds are picked up by a microphone 2, processed by a signal processor 9 and are presented to a hearing device user 10 . . . . The signal processing is based on audio processing parameters.").

203. A POSA would have understood that during an audiological fitting, multiple hearing aid profiles are programmed into the memory of the hearing aid, and during use a selection of a profile is received and the selected profile is implemented to compensate for the hearing impairment of the user. For example, Mangold discloses a plurality of "programs" that include a collection of parameters and/or coefficients in a hearing aid program for processing audio signals. Mangold, Ex. 1007 at 2:28-33, 4:47-58, 1:17-22, 1:23-29. During an audiological fitting, the programs are stored in memory and adjusted to correct for the hearing loss of a particular hearing aid user. *Id*. at Abstract, 2:3-11, 1:17-22. Mangold discloses that the remote control receives a selection of a program from a hearing aid user or in response to the current sound environment of the hearing aid user.

Id. at 3:13-15 ("A manual program control switch 18 is provided for the user of the device to select from among the several programming options stored in memory 16"), 3:40-41, 3:49-56, 3:60-66, Fig. 3, 4:11-26, 1:40-49, 1:17-22. The selected program is transmitted from the remote control to the hearing aid to be implemented. *Id*. at 3:46-49, 3:57-59, 3:60-66, Fig. 3, 4:11-26. Thus, Mangold discloses receiving a selection of a hearing aid profile and providing the selected hearing aid profile to the hearing aid.

204. It would have been obvious to modify Fichtl in view of Mangold to receive a selection of a hearing aid profile and provide the selected hearing aid profile to the hearing aid for at least the reasons discussed above with respect to element 10 – preamble. For example, it would have been obvious for Fichtl's memory 7 for storing APP values to store a plurality of hearing aid profiles and Fichtl's controller 6 to receive a selection of a stored hearing aid profile and provide the selected hearing aid profile to the hearing aid. The selection of a hearing aid profile could be, for example, in response to the current sound environment of the hearing aid user, as taught by Mangold. Mangold, Ex. 1007 at 1:40-49 (disclosing "environmentally selected events, such as selection of settings, parameters, or algorithms, where such selection is based on an automatic computation in response to the current sound environment of the wearer"), 3:49-66, Fig. 3. Providing a plurality of hearing aid profiles would allow, for example,

a profile to be selected based on the hearing aid user's current sound environment to better compensate for hearing loss in that environment.

XVI. CLAIMS 11-12 AND 15 ARE OBVIOUS OVER FICHTL (EX. 1003) IN VIEW OF MANGOLD (EX. 1007) AND SACHA (EX. 1004)

205. As provided in my detailed analysis below, it is my opinion that claims 11, 12, and 15 of the '999 patent are rendered obvious by Fichtl in view of Mangold and Sacha.

A. [Claim 11.1] "computing device of claim 10, wherein the memory stores further instructions that, when executed by the processor, cause the processor to: initiate a timer to determine the [sic, a] period of time"

206. Fichtl in view of Sacha discloses this limitation. Sacha discloses an acclimatization program and a time-based trigger for sequencing through intermediate, "sub-optimal" signal processing parameters. Sacha, Ex. 1004 at Abstract, [0012], Fig. 2 (230), [0004] ("The device may be programmed to select a signal processing parameter set … in a defined sequence based upon elapsed operating time intervals as measured by a timer"), [0015]. Sacha uses a timer 230 that "operates when the device is powered on" to "record[] the time during which the device is powered up" to maintain a "running total of the operating time" that allows the hearing aid to switch to the next parameter set after a "specified operating time interval has elapsed." *Id.* at [0015]. Thus, Sacha discloses initiating a timer to determine an amount of time during which a hearing correction

filter of the sequence is applied. Fichtl's controller in view of Sacha would initiate the timer to determine a period of time so the controller would know when to send triggers initiating application of the subsequent hearing correction filters, as discussed above with respect to elements 10.4.2 and 10.4.3.

207. It would have been obvious to modify Fichtl in view of Sacha to include Sacha's timer to allow the second hearing correction filter to be triggered after applying the first hearing correction filter for a predetermined amount of time, as taught by Sacha, instead of or in addition to in response to the hearing aid user turning the hearing aid off then on. Fichtl itself contemplates applying the next APP value before the next power-on event. Fichtl, Ex. 1003 at 4:20-24 ("it does not matter if the change due to the acclimatization algorithm is already applied during the current usage period, or . . . not until the hearing device 1 is switched off and on again"). A POSA would recognize that power-on events and operating time are each common, alternative mechanisms for triggering the next incremental correction during an acclimatization process. See Sacha, Ex. 1004 at [0004] ("The device may be programmed to select a signal processing parameter set for specifying to the signal processing circuit from a group of such parameter sets in a defined sequence based upon elapsed operating time intervals as measured by a timer or upon a specified number of detected power events representing the device being turned on."), [0013]-[0015]. Such a modification prevents the

acclimatization process from occurring too quickly, if the hearing aid user powers on and off frequently, or too slowly, if the hearing aid user powers on and off infrequently. *See* Fichtl, Ex. 1003 at 2:13-18, 1:57-2:10. Triggering subsequent incremental corrections based on time intervals also allows control over which corrections are applied for longer time intervals, such as when the hearing aid user first starts wearing the hearing aid, and which are applied for shorter time intervals, such as when the corrections are closer to the target hearing compensation. *See* Sacha, Ex. 1004 at [0015] (disclosing different time intervals for the sequence of incremental corrections).

208. Claim 11 is silent regarding where the timer is located, and therefore the timer need not be located in the "computing device." Nevertheless, to the extent one would argue that the timer must be in the "computing device," this claim would be obvious over the combination of Fichtl, Mangold, and Sacha as discussed herein, and further in view of Janssen (Ex. 1005), which discloses a computing device comprising a timer. Janssen discloses "a hearing aid that automatically adjusts itself in time" (Janssen, Ex. 1005 at [0007]) to provide acclimatization. *Id*. at [0002]-[0007]. The hearing aid includes "a seventh means for setting of the repeat interval of the successive trigger signals provided by the time clock." *Id*. at [0007]. In a preferred embodiment, the "seventh means are incorporated in an external unit." *Id*. at [0021]. It would have been obvious to

incorporate the timer into the remote control for the same reasons discussed above with respect to element 10 – preamble for incorporating components of the hearing device into a remote control. *See*, *e.g.*, Mangold, Ex. 1007 at 1:59-2:2, 4:21-26.

L. [Claim 11.2] "iteratively select and provide selection signals related to subsequent ones of the incremental hearing correction filters from the sequence to the hearing aid when the period of time exceeds the [sic, a] threshold time increment"

209. As discussed above with respect to claim 13, Fichtl in view of Mangold discloses selecting and providing selection signals to the hearing aid that comprise triggers to initiate application of the subsequent incremental hearing correction filter, or initial hearing correction filter if at the start of acclimatization, by the hearing aid. As discussed above with respect to element 11.1, Fichtl in view of Sacha discloses sending such trigger signals after the current hearing correction filter has been applied for a predetermined amount of time. Thus, Fichtl in view of Mangold and Sacha discloses iteratively selecting and providing selection signals related to subsequent ones of the incremental hearing correction filters from the sequence to the hearing aid when the period of time exceeds a threshold time increment.

M. [Claim 11.3] "reset and restart the timer when each of the subsequent ones of the incremental hearing correction filters is provided to the hearing aid"

210. Fichtl in view of Sacha discloses this limitation. Sacha discloses that the "progression from each parameter set to another may occur after the same

operating time interval, or different operating time intervals may be defined for each parameter set." Sacha, Ex. 1004 at [0015]. A POSA would have understood the progression of parameter sets using "the same operating time interval" to include a reset and restart of the timer so that the next operating time interval (and incremental hearing correction filter) would have the same operating time interval.

N. [Claim 12] "computing device of claim 10, wherein the threshold time increment varies with each of the incremental hearing correction filters"

211. Fichtl in view of Mangold and Sacha renders obvious this claim. Sacha discloses that the "progression from each parameter set to another may occur after the same operating time interval, or different operating time intervals may be defined for each parameter set." Sacha, Ex. 1004 at [0015]. Thus, Fichtl's incremental hearing correction filters could, using Sacha's timer as discussed above with respect to element 11.1, each be applied for a "different operating time interval," as taught by Sacha, in which case their corresponding threshold time increments would vary, as recited in claim 12.

O. [Claim 15] "computing device of claim 10, wherein the memory further comprises instructions that, when executed by the processor, cause the processor to progressively advance through the sequence of the incremental hearing correction filters by providing each of the incremental hearing correction filter to the hearing aid, one at a time, over a sequence of time increments to provide a progressive hearing aid adjustment from an uncompensated hearing level to a corrected hearing level to aid in the user in acclimating to the hearing aid"

212. Fichtl in view of Mangold and Sacha renders obvious this claim. As described above with respect to element 10.2 through element 10.4.3, Fichtl's memory stores instructions that cause Fichtl's controller, corresponding to the claimed processor, to progressively advance through the sequence of the incremental hearing correction filters by providing each of the incremental hearing correction filter, one at a time, to provide a progressive hearing aid adjustment from an uncompensated hearing level to a corrected hearing level to aid in the user in acclimating to the hearing aid. Fichtl further discloses time increments corresponding to the time between application of each incremental hearing correction filter (e.g., the time increment between time B and time G of Fig. 2 during which the first incremental hearing correction filter is applied). Fichtl, Ex. 1003 at Fig. 2, 3:49-53, 4:5-7, 4:26-51. In addition, as discussed above with respect to element 11.2, Fichtl in view of Sacha discloses applying each incremental hearing correction filter for a corresponding time increment.

P. A POSA Would Have Combined Fichtl, Mangold, and Sacha

213. As discussed above, Fichtl, Mangold, and Sacha disclose hearing aids capable of implementing a sequence of hearing correction filters as part of an acclimatization program, and external computing devices that communicate information, such as filters and triggers, to the hearing aids. The sequence of

hearing correction filters applied in all of the prior art references can be based on time triggers generated using a timer.

214. The combination would have involved nothing more than combining prior art elements according to known methods, such as including a timer in an external unit or in the hearing aid, with no change in their respective functions, to yield predictable results.

XVII. CONCLUSION

215. For my efforts in connection with the preparation of this declaration, I have been compensated at my standard hourly rate for this type of consulting activity. My compensation is in no way contingent on the results of these or any other proceedings relating to the above-captioned patent.

216. 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 by me with the knowledge that willful false statements are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code, and with the knowledge that such willful false statements may jeopardize the results of these proceedings.

217. I declare under penalty of perjury that the foregoing is true and correct.

218. In signing this declaration, I understand that the declaration will be filed as evidence in a review proceeding before the Patent Trial and Appeal Board of the U.S. Patent and Trademark Office. I acknowledge 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 undertake to appear for cross-examination within the United States during the time allotted for cross-examination, and within the limits of my ability so to do.

Dated: 1/26/2017

Les Atlas

Les Atlas, Ph.D.

APPENDIX A

Curriculum Vitæ

Les E. Atlas, Ph.D.

Work Address:

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University of Washington
Box 352500
Seattle, Washington 98195-2500
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Research Interests:

Theory of dynamical systems' signal processing, statistics, machine learning, and modulation frequency, with applications in time series, acoustics, machine monitoring, sensor arrays, speech, and hearing.

Professional Positions:

Professor of Electrical Engineering and Adjunct Professor of Computer Science and Engineering, University of Washington, Sept. 1994 - Sept. 1998, Sept. 2001 - present.

Visiting Scholar, Centre for the Neural Basis of Hearing, Cambridge University, Cambridge, UK, Feb. 2004 - June 2004.

Visiting Senior Scientist and Fulbright Senior Scholar, Fraunhofer Institute IIS, Erlangen, Germany, Aug. 2003 – Jan. 2004. (Taught UW Licensed technology: UW Patent, US 7136418 B2, which had a major impact on modern audio coding standards, such as AACplus and now is referenced by 33 other patents.)

Associate Chair for Research, Professor of Electrical Engineering, and Adjunct Professor of Computer Science and Engineering, University of Washington, Sept. 1998 – Sept. 2001.

Distinguished Visitor, Queensland University of Technology, Department of Electrical Engineering, Queensland, Australia, January 1994 – Sept 1994. (Most notable outcome: Helped the Commonwealth Scientific and Industrial Research Organisation, the federal government agency for scientific research in Australia, develop technology for clean coal mining via *in situ* acoustic monitoring of proximity of coal/rock interface. This technology saves the State of Queensland over AU$100M per year.)

Associate Professor of Electrical Engineering and Adjunct Associate Professor of Computer Science and Engineering, University of Washington, Sept. 1988 - Sept. 1994.

Assistant Professor of Electrical Engineering, University of Washington, Jan. 1984 - Sept. 1988.

Part-time member of technical staff, SRI International, Feb. 1982 - Nov. 1983.

Research Affiliate, Stanford University, Sept. 1981 - Dec. 1983.

Research Assistant, Stanford University, July 1978 - Sept. 1982.

Education:

University	Dates	Degree
Stanford	9/78 - 11/83	Ph.D. (in Elec. Eng.)
Stanford	9/77 - 9/78	M.S. (in Elec. Eng.)
University of Wisconsin	9/73 - 5/77	B.S.E.E. (high honors)

Honors and Awards:

- Invited to be Plenary Speaker, Conference on Implantable Auditory Prostheses, Lake Tahoe, 2015.
- Keynote Speaker, Army Research Office Workshop on RF Sensing, Arlington, 2015.
- Bloedel Scholar Award, W. Merrill Bloedel Hearing and Speech Institute, 2012-15
- Chair's Award, University of Washington Electrical Engineering, 2011.
- Nominated as IEEE Distinguished Lecturer by the Signal Processing Theory and Methods Committee of the IEEE Signal Processing Society, Nov. 2011.
- Presented an invited tutorial on Modulation Frequency and Superposition at the large (1800 attendees) IEEE ICASSP 2008 Conference, Las Vegas, NV, April 30, 2008.
- Presented an invited tutorial on Modulation Frequency at the large (1200 attendees) INTERSPEECH 2007 Conference, Antwerp, Belgium, August 27, 2007.
- Mathematics community citations of our theoretical work started a new node on the Mathematics Genealogy Project, 2007.
- Keynote Speaker, West Coast Port Security Workshop, January 13, 2007.
- Fulbright Senior Scholar (with study in Germany), 2003-4.
- Fellow of the IEEE, "For contributions to time-varying spectral analysis and acoustical signal processing," 2004.
- Teacher-of-the-Year, University of Washington Electrical Engineering 2003.
- Department Service Award, University of Washington Electrical Engineering 2003.
- Elected Member-at-Large, IEEE Signal Processing Society Board of Governors, 2003.
- Plenary Speaker, Speech Dynamics by Ear, Eye, Mouth And Machine, An Interdisciplinary Workshop Organized by The Institute for Electronics, Information and Communication Engineers (IEICE) and The Acoustical Society of Japan., Kyoto, Japan, June 27, 2003.
- Keynote Speaker, Conf. On Advanced Signal Signal Processing Algorithms, Architectures, and Implementations, Int. Symp. On Optical Science and Technology, July, 2001, San Diego.
- One of ten experts' summary speakers, IEEE International Conference on Acoustics, Speech, and Signal Processing, June 2000, Istanbul, Turkey.
- Invited as Plenary Speaker, Symposium 2000 on Adaptive Systems for Signal Processing, Communications and Control, Alberta, Canada, October 2000.
- One of ten experts' summary speakers, IEEE International Conference on Acoustics, Speech, and Signal Processing, March 1999, Phoenix.
- General Chairman of the 1998 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP '98), which brought 2,200 participants to Seattle and was the largest ICASSP event ever, May 1998.
- Invited to be Keynote Speaker at the International Symposium on Industrial Electronics,

Portugal, 1997. (Declined due to illness.)

- Best Paper Award, 1997: McLaughlin, J., L. Atlas, L. Owsley, and G.D. Bernard, "Advances in Real-Time Monitoring of Acoustic Emissions," *Proceedings of the 1997 SAE Aerospace Manufacturing Technology Conference*, Seattle, WA, June, 1997.
- Keynote Speaker, International Symposium - Anthropomorphic Systems of Automatic Speech Recognition and Synthesis, St. Petersburg, Russia, June 30-July 2, 1993.
- Nominated as IEEE Distinguished Lecturer by the Neural Network Committee of the IEEE Signal Processing Society, Nov. 1992.
- General Chairman, IEEE International Symposium on Time-Frequency and Time-Scale Analysis, Victoria, BC, Oct. 4-6, 1992.
- Keynote Speaker, Phillips Doppler Ultrasound Conference, June 12, 1991.
- Chosen as one of the 100 Seattle residents under 40 who will be "...shaping the 1990's." (*Seattle Weekly*, July 4, 1989).
- University of Washington, College of Engineering Nominee for NSF Waterman Prize, 1987.
- National Science Foundation Presidential Young Investigator Award, 1985 - 1989.
- Physio-Control Career Development Award, 1984-87.

- Numerous Awards Received by Undergraduate Students, including, most recently:
    - 2 Mary Gates Leadership Awards. (As President of UW Husky Cycling Club, under Atlas mentorship.)
    - 1 Luce Scholar Award.
    - 4 Mary Gates Research Awards.

Publications:

Top Cited and Wide-Impact Papers (Citation numbers from Google Scholar, November 25, 2015)

1. D Cohn, L Atlas, and R Ladner, "Improving generalization with active learning," *Machine Learning* **15** (2), pp. 201-221, 1994. (1096 citations)
   (Started the machine learning field of active learning.)
2. DC Park, MA El-Sharkawi, RJ Marks, LE Atlas, and MJ Damborg, "Electric load forecasting using an artificial neural network," *IEEE Transactions on Power Systems,* **6** (2), 442-449, 1991. (910 citations) (Started the use of artificial neural network technology for power system forecasting.)
3. JT Connor, RD Martin, and LE Atlas, "Improving performance in noise for hearing aids and cochlear implants using coherent modulation filtering," *IEEE Transactions on Neural Networks,* **5** (2), pp. 240-254, 1994. (545 citations) (Started the statistical analysis of the now-common recurrent neural networks.)
4. J Y Zhao, LE Atlas, RJ Marks, "The use of cone-shaped kernels for generalized time-frequency representations of nonstationary signals," *IEEE Transactions on Acoustics, Speech and Signal Processing*, 38 (7), pp. 1084-1091, 1990. (442 citations) (Solved the open problem of retaining the Wigner distribution's time resolution while reducing its quadratic interference terms. Now used within a wide range of DSP monitoring systems and products worldwide.)

Refereed Archival Journal Papers

5. Atlas, L., M. Herndon, F. Simmons, L. Dent, and R. White, "Results of Stimulus and Speech

Coding Schemes Applied to Multi-Channel Electrodes," *Annals of the New York Academy of Science,* **405,** pp. 377-386, 1983.
6. Simmons, F., M. Herndon, L. Atlas, R. White, and L. Dent, "Multielectrode Modular Simulation: Some Selected Psycho-physical and Speech Results," *Advances in Audiology,* **2,** pp. 163-169, 1984.
7. Marks, R. and L. Atlas, "Composite Matched Filtering with Error Correction," *Optics Letters,* **12,** pp. 135-137, 1987.
8. Atlas, L., "Auditory Coding in Higher Centers of the CNS," *IEEE Engineering in Medicine and Biology Magazine,* **6,** pp. 29-32, June 1987. (Invited)
9. Marks, R., J. Ritcey, L. Atlas, and K. Cheung, "Composite Matched Filter Output Partitioning," *Applied Optics,* **26,** pp. 2274-2278, 1987.
10. Cheung, K.F., L. Atlas, J. Ritcey, C. Green, and R. Marks, "Conventional and Composite Matched Filters with Error Correction: A Comparison," *Applied Optics,* **26,** pp. 4235-4239, 1987.
11. Cheung, K.F., L. Atlas, and R. Marks, "Synchronous vs. Asynchronous Behavior of Hopfield's CAM Neural Net," *Applied Optics,* **26,** pp. 4808-4813, 1987.
12. Marks, R., L. Atlas, J. Choi, S. Oh, K. Cheung and D. Park, "A Performance Analysis of Associative Memories with Nonlinearities in the Correlation Domain," *Applied Optics,* **27,** pp. 2900-2904, 1987.
13. Oh, S., D. Park, R. Marks, and L. Atlas, "Error Detection and Correction in Multilevel Algebraic Processors," *Optical Engineering,* **27,** pp. 289-294, 1988 (Invited).
14. Marks, R., L. Atlas, and K. Cheung, "Optical Processor Architectures for a Class of Continuous Level Neural Nets," *Optics Letters,* **13,** pp. 533-535, 1988.
15. Cheung, K.F., R.J. Marks and L. Atlas, "Convergence of Howard's Minimum-Negativity-Constraint Extrapolation Algorithm," *JOSA-A,* Vol. 5, No. 11, pp. 2008-2009, Nov. 1988.
16. Oh, S., D.C. Park, R.J. Marks and L.E. Atlas, "Nondispersive Propagation Skew in Iterative Neural Network and Optical Feedback Processors," *Optical Engineering,* Vol. 28, No. 5, pp. 526-532, May 1989.
17. Marks, R.J., S. Oh and L. Atlas, "Alternating Projection Neural Networks," *IEEE Transactions on Circuits and Systems,* Vol. 36, No. 6, pp. 846-857, June 1989.
18. Marks, R.J., L.E. Atlas, S. Oh and K.F. Cheung, "Optical-Processor Architectures for Alternating-Projection Neural Networks," *Optics Letters,* Vol. 13, No. 6, pp. 533-535, June 1988.
19. Marks, R.J., L.E. Atlas, J.J. Choi, S. Oh, K.F. Cheung and D.C. Park, "Performance Analysis of Associative Memories with Nonlinearities in the Correlation Domain," *Applied Optics,* Vol. 27, No. 14, July 1988.
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97. Atlas, L., Li, Q., and Thompson, J., "Homomorphic Modulation Spectra," *Proc. IEEE*

ICASSP '04, Montreal, CA, May, 2004.

98. Schimmel, S. & Atlas, L., "Coherent Envelope Detection for Modulation Filtering of Speech," *Proc. IEEE ICASSP '05*, Philadelphia, PA, March, 2005.
99. Li, Q. & Atlas, L., "Properties for Modulation Spectral Filtering," *Proc. IEEE ICASSP '05*, Philadelphia, PA, March, 2005.
100. Atlas, L. & Janssen, C., "Coherent Modulation Spectral Filtering for Single-Channel Music Source Separation," *Proc. IEEE ICASSP '05*, Philadelphia, PA, March, 2005.
101. Owsley, L., Atlas, L. & Heinemann, C., "Use of Modulation Spectra for Representation and Classification of Acoustic Transients from Sniper Fire," *Proc. IEEE ICASSP '05*, Philadelphia, PA, March, 2005.
102. Sukittanon, S. Atlas, L., Pitton, J., and Filali, K., "Improved Modulation Spectrum through Multi-Scale Modulation Frequency Decomposition," *Proc. IEEE ICASSP '05*, Philadelphia, PA, March, 2005.
103. Schimmel, S. and L. Atlas, "Generalization of Modulation Filtering to Non-Uniform Sub-Bands," *Proceedings of the SPIE - The International Society for Optical Engineering*, Denver, CO, August, 2005. (Invited)
104. Takahiro Shinozaki, Mari Ostendorf, Les E. Atlas, "Data sampling for improved speech recognizer training," *Proc. Interspeech 2005*, pp. 1693-1696.
105. Schimmel, S., Fitz, K., and Atlas, L., "Frequency Reassignment for Coherent Modulation Filtering," *Proc. IEEE ICASSP '06*, Toulouse, France, May, 2006.
106. Sukittanon, S., Atlas, L., Pitton, J., and Filali, K., "Enhanced Modulation Spectrum Using Space-Time Averaging for In-Building Acoustic Signature Identification," *Proc. IEEE ICASSP '06*, Toulouse, France, May, 2006.
107. Atlas, L., B. Atal, and P. Clark, "Modulation decompositions for broadband signals," *Proceedings of the SPIE - The International Society for Optical Engineering*, San Diego, CA, August, 2006. (Invited)
108. Smith, B., Atlas, L., and Gupta, M., "Beamforming Alternatives for Multi-Channel Transient Acoustic Event Classification," *Proc. IEEE ICASSP '07*, Honolulu, HI, April, 2007, pp. II-993 – II-996.
109. Schimmel, S., L. Atlas, and K. Nie, "Feasibility of Single Channel Speaker Separation Based on Modulation Frequency Analysis," *Proc. IEEE ICASSP '07*, Honolulu, HI, April, 2007, pp. IV-605 – IV-608.
110. K. Nie, L. Atlas, B. King, J. Rubinstein, "Speech Recognition with Coherent Envelopes," Conference on Implanted Auditory Prostheses, Lake Tahoe, CA, July, 2007
111. Patrick McVittie and Les Atlas, "Air-Coupled Ultrasound Time-Of-Flight Estimation for Shipping Container Cargo Verification," *Proc. IEEE ICASSP '08*, Las Vegas NV, March 30 - April 4, 2008, pp. 1797 – 1800.
112. Stephen D. Hawley and Les E. Atlas and Howard J. Chizeck, "Some Properties of An Empirical Mode Type Signal Decomposition Algorithm," *Proc. IEEE ICASSP '08*, Las Vegas NV, March 30 - April 4, 2008, pp. 3625 – 3628.
113. Pascal Clark and Les Atlas, "Modulation Decompositions for the Interpolation of Long Gaps in Acoustic Signals," *Proc. IEEE ICASSP '08*, Las Vegas NV, March 30 - April 4, 2008, pp. 3741 – 3744.
114. Schimmel, S. and L. Atlas, "Target Talker Enhancement in Hearing Devices," *Proc. IEEE ICASSP '08*, Las Vegas NV, March 30 - April 4, 2008, pp. 4201 – 4204.
115. Kaibao Nie, Les Atlas, and Jay Rubinstein, "Single Sideband Encoder for Music Coding in

Cochlear Implants," *Proc. IEEE ICASSP '08*, Las Vegas NV, March 30 - April 4, 2008, pp. 4209 – 4212.

116. Qin Li and Les Atlas, "Coherent Modulation Filtering for Speech," *Proc. IEEE ICASSP '08*, Las Vegas NV, March 30 - April 4, 2008, pp. 4481 – 4484.
117. Brian King and Les Atlas, "Coherent Modulation Comb Filtering for Enhancing Speech in Wind Noise," *Proc. of the 11th International Workshop on Acoustic Echo and Noise Control*, Seattle, Washington USA, 14–17 September 2008.
118. P. Clark and L. Atlas, "A sum-of-products model for effective coherent modulation filtering," *Proc. IEEE ICASSP*, Taipei, Taiwan, 2009.
119. K. Nie, X. Li, L. Atlas, and J. Rubinstein, "Experimental results with a single sideband encoding strategy for cochlear implants," *Conference on Implantable Auditory Prostheses*, Lake Tahoe, CA, July 12-17, 2009.
120. B. King and L. Atlas, "Single-channel Source Separation using Simplified-training Complex Matrix Factorization," *Proc. IEEE ICASSP*, Dallas, TX, 2010.
121. X. Li, K. Nie, L. Atlas, and J. Rubinstein, "Harmonic coherent demodulation for improving sound coding in cochlear implants," *Proc. IEEE ICASSP*, Dallas, TX, 2010.
122. P. Clark, I. Kirsteins and L. Atlas, "Multiband analysis for colored amplitude-modulated ship noise," *Proc. IEEE ICASSP*, Dallas, TX, 2010.
123. Pascal Clark, Gregory Sell, and Les Atlas "A Novel Approach using Modulation Features for Multiphone-Based Speech Recognition," *Proc. IEEE ICASSP*, Prague, Czech Republic, May 22-27, 2011.
124. "Speech Recognition with Segmental Conditional Random Fields: A Summary of the JHU CLSP 2010 Summer Workshop," *Proc. IEEE ICASSP*, Prague, Czech Republic, May 22-27, 2011.
125. J. Froehlich, E.C. Larson, E. Saba, T. Campbell, L. Atlas, J. Fogarty, S.N. Patel. "A Longitudinal Study of Pressure Sensing to Infer Real-World Water Usage Events in the Home," *Proceedings of the Ninth International Conference on Pervasive Computing*. San Francisco, California. June 12-15, 2011.
126. P. Clark, I.P. Kirsteins, and L. Atlas, "Existence and estimation of impropriety in real rhythmic signals," *Proc. IEEE ICASSP*, Kyoto, Japan, 2012.
127. P. Clark, I. Kirsteins, and L. Atlas, "Complementary Envelope Estimation for Frequency-Modulated Random Signals", *Proc. IEEE ICASSP*, Vancouver, Canada: 2013.
128. T. Powers, L. Atlas, E. Hanusa, and D. W. Krout, "Track to Track Fusion: PACsim Data Set" 17th International Conference on Information Fusion, Salamanca, Spain, 2014.
129. S. Wisdom, T. Powers, L. Atlas, and J. Pitton, "Enhancement of Reverberant and Noisy Speech by Extending Its Coherence," in *Proc. REVERB Challenge Workshop*, Florence, Italy, May 2014.
130. S. Wisdom, L. Atlas, and J. Pitton, "Extending Coherence Time for Analysis of Modulated Random Processes", *Proc. IEEE ICASSP*, Florence, Italy, May 2014.
131. S. Wisdom, J. Pitton, and L. Atlas, "Extending Coherence for Optimal Detection of Nonstationary Harmonic Signals," *Proc. Asilomar Conference on Signals, Systems, and Computers*, Pacific Grove, CA, Nov. 2014.
132. D. W. Krout and T. Powers, "Sensor Management for Multistatics" *Proc. 17th International Conference on Information Fusion*, Salamanca, Spain, 2014.
133. B. Ekin and L. Atlas, "Modern Data Communication Theory Suggests Processing for EEG Signals", *Proc. 2014 IEEE Engineering in Medicine and Biology Society BRAIN Grand*

Challenge Conference, Washington, DC, 13-14 November 2014.
134. Okopal, S. Wisdom, and L. Atlas, "Estimating the Noncircularity of Latent Components within Complex-Valued Subband Mixtures with Applications to Speech Processing," *Proc. Asilomar Conference on Signals, Systems, and Computers*, Pacific Grove, CA, Nov. 2014.
135. S. Wisdom, G. Okopal, L. Atlas, and J. Pitton, "Voice Activity Detection Using Subband Noncircularity," *Proc. IEEE ICASSP*, Brisbane, Austrailia, April 2015.
136. T. Powers, D.W. Krout, L. Atlas, "Sensor Selection from Independence Graphs Using Submodularity" *Proc. 18th International Conference on Information Fusion*, Washington DC, USA, July 2015.

Published Abstracts:

1. Atlas, L., M. Herndon, R. White and L. Dent, "Development of a Portable Speech Processor for an Auditory Implant Subject," *J. Acoust. Soc. Am. Suppl. 1,* **71,** p. S57, Spring 1982.
2. Herndon, M., L. Atlas, F. Simmons and R. White, "Comparison of four Speech Processing Algorithms in One Cochlear Implant Subject," *J. Acoust. Soc. Am. Suppl. 1,* **71,** p. S57, Spring 1982.
3. Atlas, L., and A. Clayton, "Coherence-Based Enhancement of Noisy Speech," *J. Acoust. Soc. Am. Suppl. 1,* **80,** p. S20, Fall 1986.
4. Atlas, L., J. Ritcey, K. Cheung and R. Marks, "Improving the Performance of Composite Matched Filters," Digest of Tech. Papers for Annual Meeting of Optical Soc. of Am., p. 31, Oct. 1986.
5. Green, C., K. Cheung, L. Atlas and R. Marks, "Performance of Conventional and Composite Matched Filters with Error Correction," Digest of Tech. Papers for Annual Meeting of Optical Soc. of Am., p. 31, Oct. 1986.
6. McArthur, G., C. Schmid, M. Warnecke, and L. Atlas, "The Generation of Propeller Modulation Sounds for Sonar Trainers," *J. Acoust. Soc. Am., Suppl. 1,* **82,** p. S86, Spring 1987.
7. Zhao, Y., L. Atlas and R. Marks, "An Optimal Class of Generalized Time-Frequency Representations for Nonstationary Signal Analysis," *J. Acoust. Soc. Am., Suppl. 1,* **83,** p. S1, Spring 1988.
8. Schlatter, J. and L. Atlas, "A Real-Time Frequency-Shift Decoder for the Last Transmission of Korean Airlines Flight 007," *J. Acoust. Soc. Am., Suppl. 1,* **83,** P. S2, Spring 1988.
9. Pope, C., L. Atlas, and C. Nelson, "An Artificial Neural Network-Based Codebook Search Technique," *J. Acoust. Soc. Am., Suppl. 1,* **83,** p. S53, Spring 1988.
10. Atlas, L., R. Marks, and J. Taylor, "Network Learning Modifications for Multi-Modal Classification Problems: Applications to EKG Patterns," *1st Meeting of International Neural Network Society,* Boston, MA, Sept. 6-10, 1988.
11. Loughlin, P. J., J.W. Pitton and L.E. Atlas, "A New Time-Frequency Representation for Speech and Range-Doppler Processing," *1990 Digital Signal Processing Workshop,* Mohonk Mountain House, New Paltz, New York, Sept. 16-19, 1990.
12. Riskin, E. A., R. Ladner, L.E. Atlas, and R-Y. Wang, "Index Reassignment for Progressive Transmission of Full Search Vector Quantization," *Abstracts of Papers of 1993 IEEE International Symposium on Information Theory,* p. 173, January 1993.
13. Pitton, J., E. Burns, and L. E. Atlas, "Detection of Amplitude Modulation with Linear-frequency-modulated Carriers," *J. Acoust. Soc. Am.,* **Vol. 93**, No. 4, Pt. 2, p. 2314, April, 1993.
14. Atlas L.E., J. Pitton, and P. Loughlin, "Proper Time-frequency Distributions for Speech," *J.*

*Acoust. Soc. Am., **Vol 93***, No 4, Pt. 2, p. 2318, April, 1993.

15. L. Atlas, "Adapting time-frequency responses to speech and acoustic classification tasks," *J. Acous. Soc. Am.,* 106(4), October 1999.
16. Souza, P., Atlas, L., Schimmel, S., Rubenstein, J., Drennan, W., and Won, J.-H., "Coherent modulation enhancement: Improving performance in noise for hearing aids and cochlear implants," *J. Acous. Soc. Am.*, April 2005, Vol. 117, Issue 4, p. 2535.
17. S. Philips, J. Pitton, and L. Atlas, "Automatic feature identification using multidimensional scaling data," *J. Acous. Soc. Am.*, June 2006.
18. Nie, K., Atlas, L., Clark, P., Atal, B., and Rubinstein, J., "Effective Envelope Modification by Coherent Demodulation," *Abstracts of the Thirtieth Annual Midwinter Research Meeting of the Association for Research in Otolaryngology*, Denver, February 10-15, 2007, page 441.

Graduate Student Dissertation and Thesis Supervision:

Principal or Co-Principal Advisor to Ph.D. Students:

1. Advisor to Bradley Ekin.
2. Advisor to Eldridge Alcantara, supported by the DoD's Science, Mathematics and Research for Transformation (SMART) Scholarship. Eldridge received the 2014 College of Engineering Award for Teaching.
3. Advisor to Thomas Powers who passed his Ph.D. General Exam.
4. Co-Advisor to Scott Wisdom, who passed his Ph.D. qualifying exam in July of 2012. His Ph.D. General Exam will be by Spring 2016.
5. Advisor to Xing Li who finished her Ph.D. late in 2012. She is now at Microsoft Research.
6. Brian King passed his Ph.D. final exam in Summer 2012. He is now at Amazon Research.
7. Charles Pascal Clark passed his Ph.D. final exam in Summer 2012. He is on the faculty at Johns Hopkins University and is on leave at Apple Computer.
8. Scott Phillips, EE, who finished his Ph.D. in August of 2007. Dr. Phillips is a member of the technical staff at MIT Lincoln Labs.
9. Steven Schimmel, EE, who finished his Ph.D. in July of 2007. Dr. Schimmel is on the faculty at the University of Zurich and is on leave at Google in Zurich.
10. Qin Li, EE, who passed his Ph.D. general exam in Fall 2004. Mr. Li is on leave, working at the audio coding group at Microsoft.
11. John Keane, EE, who finished his Ph.D. in December of 2004. Dissertation: "Detection Theory for Cell Signaling." Dr. Keane is a Staff Scientist at the Fred Hutchinson Cancer Research Center.
12. Somsak Sukittanon, EE, who finished his Ph.D. in December of 2003. Dissertation: "Audio/video match and mining." Dr. Sukittanon is currently an Associate Professor at the University of Tennessee.
13. Brad Gillespie, EE, who finished his Ph.D. in December of 2002. Thesis: "Strategies for Improving Audible Quality and recognition Accuracy of Reverberant Speech." Dr. Gillespie, who was a Microsoft Fellow, worked in future strategies as Assistant to Craig Mundie, Chief Technology Officer at Microsoft. He is now Principal in a start-up in NYC.
14. Jasha Droppo, EE, who finished his Ph.D. in June of 2000. Thesis: "Classification of Transient Events in Time Series." Dr. Droppo works as a Researcher in the Speech Technology Group at Microsoft Research.
15. Lane Owsley, EE, who finished his Ph.D. in March of 1998. Dr. Owsley is currently employed as Senior Engineer at the UW Applied Physics Lab.
16. Jack McLaughlin, EE, who finished his Ph.D. in July of 1997. Dr. McLaughlin is currently

employed as a Senior Engineer at UW Applied Physics Lab.

17. Sivabala Narayanan, EE, who finished his Ph.D. in January of 1995. Dr. Narayanan is currently employed as a Research Professor at UW's Med. School Div. of Cardiology.
18. Jing Fang, EE, who finished her Ph.D. in Dec. of 1994. Dr. Fang is at Motorola's R&D Group, Austin, TX.
19. James Pitton, EE, who finished his Ph.D. in February of 1994. Dr. Pitton is currently a division director at UW Applied Physics Lab, and finished a 4 year temporary assignment to the Office of Naval Research International Field Office in London and now manages several groups at UW Applied Physics Lab.
20. Patrick Loughlin, EE, who finished his Ph.D. in October of 1992. He is the William Kepler Whiteford Professor (and formerly Chair) of the Electrical Engineering and Bioengineering Departments at the University of Pittsburgh. Prof. Loughlin is a Fellow of the IEEE.
21. Jerome T. Connor, EE, who finished his Ph.D. in Feb. of 1993. Dissertation Title: "Time series and neural network modeling," Dr. Connor is a Professor at the London School of Business.
22. Co-advisor to David Cohn, a Ph.D. student in the Department of Computer Science and Engineering, who finished his Ph.D. in 1993. Dr. Cohn worked for Google, Inc, started their machine learning effort, greatly improving AdSense with the active learning concepts, which we first initiated during his Ph.D. He retired from Google after his stock vested, restores a fleet of WW II military aircraft and does volunteer computer system management at the South Pole in Antarctica.
23. Yunxin Zhao, EE, who finished her Ph.D. in Feb. of 1988, Dissertation: "On Stochastic Modeling and Time-Frequency Representation of Signals with Application to Speech Recognition and Image Restoration." She is a Professor of Computer Science at the University of Missouri and a Fellow of the IEEE.

M.S.E.E. Thesis and Project Students:

1. Tyler Ganter, EE, MSEE June 2015.
2. Brian King, EE, MSEE March 2007.
3. Pascal Clark, EE, MSEE December 2007. Mr. Clark's MSEE Thesis "Distortion-Free Coherent Modulation Filtering and Interpolation of Long Gaps in Acoustic Signals" won the University of Washington's Graduate School's Distinguished Thesis Award
4. Cameron Colpitts, EE, MSEE March 2007. Cameron is employed in financial consulting.
5. Dan Luong, EE, MSEE December 2007. Danny works for Raytheon in San Diego.
6. Brandon Smith, EE, MSEE awarded June 2006. He is employed at Neural Audio, Kirkland, WA.
7. Travis Wilkins, EE, MSEE awarded June 2005. He is employed at Intel, Beaverton, OR.
8. Jeffrey Thompson, EE, audio coding. June 2003. After an internship at Fraunhofer Institute, Erlangen, Germany (within the MP3 group), He is employed at Neural Audio, Kirkland, WA.
9. Qin Li, EE, periodicity estimation and speech coding, June 2003. He continued on with me for Ph.D. studies and is currently on leave at Microsoft.
10. Mark Vinton, EE, audio coding. July 2001. He is currently at Dolby Labs, San Francisco. *Mark won a 2004 Emmy Award for Outstanding Achievement in Engineering*.
11. Alan Gibbs, 1998, Thesis: Analysis-Synthesis with the Short-Time Zak Transform.
12. James Droppo, 1996, Thesis: Signal Processing and Segmentation for an Acoustic Pyrometer. James continued on for a Ph.D. with me.
13. Lane Owsley, 1994, Thesis: Ordered Vector Quantization for Pattern Recognition. Lane

continued on for a Ph.D. with me.

14. Roy Peterson, 1993, Thesis: Comparison of IIR Initialization Techniques for Improved Color Doppler Wall Filter Performance. Roy is now VP of Engineering at Phillips Ultrasound.
15. Rod Mercer 1994, MSEE Project on Software Systems for Time-Frequency Analysis.
16. Laura Johnston, 1992, Thesis: Class-Conditional Time-Frequency Representations.
17. Erik Godo, 1992, MSEE Project on Fast Hardware for DSP.
18. Christian Roehr, 1991, Thesis: Queries for Speech Recognition.
19. Mark Donnell, 1990, Thesis: Explorations in the Use of Queries in Dynamic Time-Warp Speech Recognition.
20. Debra Rigdon, 1990, MSEE Project on Classification Trees.
21. William Kooiman, 1989, Thesis: Time-Frequency Speech Displays that are an Improvement Over the Spectrogram.
22. James Taylor, 1989, Thesis: A Comparison of Back-Propagation and Alternating Projection Neural Networks Based on R-Wave Detection in the EKG.
23. Jean McAuliffe, 1989, Thesis: A Comparative Study of Codebook Generation Techniques for Image Vector Quantization.
24. Charles Nelson, 1989, MSEE Project on Real-Time Speech Quantizer.
25. Charles Pope, 1988, Thesis: A comparison between a neural network and the generalized Lloyd algorithm for vector quantization of speech.
26. Donn Gabrielson, 1988, Thesis: An Example of the Application of Galois Fields to Sub-Sea Communications.
27. John Schlatter, 1988, Thesis: A Real Time Frequency Shift Decoder for the Last Transmission of Korean Airlines Flight 007.
28. Jim Nelson, 1987, Thesis: The Analysis of “Chirp” Source Waveforms for Shallow Water Seismic Profiling.
29. J. Michael Hill, 1987, Thesis: An Architecture for Real Time Backpropagation of Fields.
30. Rex Andrews, 1987, Thesis: An Interactive Speech Enhancement Workstation.
31. Alexandra Clayton, 1986, Thesis: Enhancement of Noisy Speech Using Coherence Processing.
32. Wei-Ping Chen, 1986, Thesis: Noise Detection in Degraded Speech Using Cross-Channel Correlation.
33. Daryoush Mehrtash, 1986, Thesis: A Comparison of a Micro-Coded VAX and TMS320 Architectures for Digital Signal Processing Applications.
34. Leo Hengky, 1985, Thesis: Speech Enhancement Using Cross-Channel Correlation.

Research Grants and Contracts:

1. Army Research Office, “Active sensing in the multisensory world — A platform for discovery in brains, computers, and their interfaces,” Multi-University Research Initiative, Co-PI, Directed by Prof. Mark Wallace, Vanderbilt, $10M including UK partners, June 2016-May 2021. (White paper approved, one of two full proposals solicited, due early December 2015.)
2. Wallace A Coulter Foundation, “Algorithm to Provide Enhanced Perception of Tonality in Cochlear Implants,” June 2014-June2016, $200,000 Direct Costs. (Principal Investigator).
3. Army Research Office, “Alternatives to Digital Processing and Demodulation for Standoff Sensors,” 2015-2016, $49,973. (Principal Investigator).
4. JSALT Workshop on Machine Learning, Speech and Natural Language Processing, 2015: $574,937 direct costs to UW, with Atlas PI. Sponsored via flow through via Johns Hopkins

from by the National Science Foundation, Google, Microsoft, Amazon, and Mitsubishi Electronics Research Lab. (Principal Investigator).

5. US Government Intelligence Community, Postdoctoral Fellow, Representations for Audio Collection, October 2012- Sept 2014, $220,000. (Principal Investigator).
6. Army Research Office, "Complementary Correlation: New Features for Audio Analysis and Enhancement," September 2012 – August 2015, $390,118. (Principal Investigator).
7. Adobe, Incorporated, "Gift for Modulation Frequency Filtering Research," $50,000 direct costs. (Principal Investigator).
8. Office of Naval Research, "Complex Modulation and Passive Sonar," October, 2007 – December 2012. $641,354. (Principal Investigator).
9. Air Force Office of Scientific Research, "Sparseness and Demodulation Signal Processing," October 2008 – December 2012. $920,241. (Principal Investigator)
10. Boeing, In Situ Monitoring for Drilling in Composites using Demodulation Signal Processing. (Co-PI, with M. Ostendorf, PI.) About $95K/year for Atlas. Joint with Prof. M Ramulu, UW Mechanical Engineering. 2010-12.
11. UW CSNE Center, Summer 2012-Spring 2014, 2 years PDRA Support for Eliott Saba, $94,292 direct costs. (Principal Investigator, with Prof. K.C. Lee, UW I-Labs)
12. Air Force Office of Scientific Research, "Coherent Modulation Frequency Representation and Filtering," Sept. 2005-July 2012, $793,110. (Principal Investigator.)
13. Office of Naval Research, "Complex Modulation and Passive Sonar," October 1, 2007 - September 30, 2012, $541,354 (Principal Investigator)
14. Boeing Corporation, Novel In-Situ Monitoring, May 2010-Jan. 2012, $215,000 for Atlas. (Co-principal Investigator, with M. Ostendorf, overall Principal Investigator)
15. <u>Multi-University Research Initiative administered by the Office of Naval Research</u>, "Center for Acoustics and Auditory Research," Apr. 1997 – Apr. 2003, $3,900,000 total (Joint with U. Maryland and Boston U. UW Principal Investigator and Overall Director of Technology Transfer.). Director and Principal Investigator.
16. Microsoft Research Gift Fund, Initiated October 1997, $100,000. (Principal Investigator).
17. Boeing (originated within DARPA Joint Strike Fighter Predictive Maintenance program), "Anomaly Detection in Complex Systems," March 1999 – June 2001, $204,000. (Principal Investigator).
18. Digital Arts, University (of Washington) Initiatives Fund, July 2001 – June 2005, $1,959,000, joint with UW Depts. of Music, Art, Architecture, and CSE. (Co-Principal Investigator).
19. Washington Research Foundation, Research Gift for "Audio and Hearing Aid Research Demonstrations," Initiated July 2001, $35,000. (Principal Investigator).
20. Fraunhofer Gesellschaft, "Joint Frequency Analysis for Audio Coding and Representation," July 2002 – June 2003, $61,000 plus UW patent and software licensing fees. (Principal Investigator).
21. DARPA, "Advanced Speech Encoding, Noise Reduction and Speaker Authentication Incorporating Non-Acoustic Measurements," June 2002-December 2004, $653,000 (Joint with UW Applied Physics Lab). (Principal Investigator).
22. Army Research Lab, "Acoustic Event Detection and Localization," Sept. 2001-June 2006, $345,000. (Principal Investigator).
23. Office of Naval Research, "Theory and Application of Modulation Frequency," Jan. 2002- December 2005, $442,075. (Principal Investigator).
24. Office of Naval Research, "Perception of Sonar," September 2002-August 2006, $435,000

total (Joint with UW APL). (Principal Investigator).

25. National Science Foundation, "A Computing Lab for Integrated Teaching of Systems Courses in Electrical Engineering," Sept. 2005-August 2008, $160,000. (Co-Principal Investigator).
26. National Science Foundation, "MSM: Adaptive Multi-Scale Model Simulation, Reduction and Integration for Cardiac Muscle Physiology," Sept. 2005-August 2008, $343,334 (Joint with UW Bioengineering). (Co-Principal Investigator).
27. Adobe Research Gift Fund, Initiated March 2006, $110,000. (Principal Investigator).
28. AT&T Research Gift Fund, Initiated May 2006, $35,000. (Principal Investigator).
29. Erudite Corporation Research Gift Fund, Initiated March 2007, $65,000. (Principal Investigator).
30. Graduate School Research Fund, "Cross-Channel Correlation to Improve the Intelligibility of Noisy Speech," Jan. 1985-June 1985, $7,124. (Principal Investigator)
31. BABECO, Inc. (An NIH-funded small business), "High-Quality Speech Synthesis," Jan. 1985 - June 1985, $19,916. (Principal Investigator)
32. Digital Equipment Corporation, Donation of Micro-VAX I for signal processing research, Aug. 1984, $18,224. (Principal Investigator)
33. National Science Foundation, Presidential Young Investigator Award, July 1985 - June 1991, $125,000 ($25,000 annually) plus $186,500 ($37,500 annually) in matching funds (fully matched). (Principal Investigator)
34. Symbolics, Inc., Signal processing research, Dec. 1985, $40,000. (Principal Investigator) (Matched first year of Presidential Young Investigator Award.)
35. Boeing Computer Services, "The Enhancement of Noisy Speech for Recognition by Computer," Jan. 1986 - Dec. 1986, $37,500. (Principal Investigator) (Matched second year of Presidential Young Investigator Award.)
36. AT&T, Donation of 3B2 Microcomputer system for research in neural networks and teaching in signal processing, Jan. 1986, $35,100. (Co-Principal Investigator)
37. Lawrence Porter Aviation Consultants, "Enhancement of Speech from KAL 007 Aviation Accident," $6,000 Gift account for equipment and direct equipment donations with a value of $2,600. (Principal Investigator)
38. Office of Naval Research, "Increasing the Accuracy of Optical Processors," July 1986 - Sept. 1989, $320,000. (Co-Investigator) ($26,000 of total budget for L. Atlas)
39. Boeing High Technology Center, "Analysis and Applications of Neural Nets," July 1986 - Aug. 1987, $118,873. (Principal Investigator) (Matched third year of Presidential Young Investigator Award.)
40. Washington Technology Center, "Neural Network Computer Architectures," Sept. 1987 - June 1990, $125,000 (Co-Principal Investigator with R. Ladner and R. Marks). (Project Coordinate and renewed for 1990-1991.)
41. Physio-Control, Inc., "Grant for Neural Network Classifier Research," March 1988, $10,000. (Principal Investigator)
42. US West Advanced Technologies, "Improving Speech Recognition with Modern Statistical Processing and Statistical Modeling Methods," June 1988 - May 1989, $100,000. (Co-Principal Investigator) (Matched fourth and fifth year of Presidential Young Investigator Award.)
43. National Science Foundation, "Power System Stability and Security Assessment Using Artificial Neural Networks," September 1988 - Sept. 1991, $377,500, (Co-Principal In-

vestigator with M. Damborg, M. El-Sharkawi and R. Marks.)

44. Puget Sound Power and Light, “Electric Load Forecasting,” Sept. 1989 - December 1990, $90,000. (Co-Principal Investigator.)
45. Office of Naval Research, “Perceptually Motivated Time-Frequency Displays,” October 1989 - September 1991, $45,000 plus $82,000 to UW Applied Physics Labs. (Co-Principal Investigator and Project Coordinator.)
46. Boeing Commercial Airplane Company, “Advanced Time-Frequency Display and Interpretation Systems,” Jan. 1990 - Dec. 1992, $436,000. (Principal Investigator.)
47. Bloedel Hearing Research Foundation, “Nonlinear transforms as a basis for peripheral auditory processing,” March 1991 - March 1993, $9,800. (Principal Investigator.)
48. National Institutes of Health, “Neural Network Assisted Cardiac Diagnosis,” July 1991 - June 1992, $120,000. (Co-Principal Investigator with Florence Sheehan.)
49. University of Washington, Royalty Research Fund, "Quadratic Signal Processing," June 1992 - Aug. 1993, $15,000. (Principal Investigator with Eve Riskin.)
50. Hughes Aircraft Company, “Neural Network Analysis of Selected Passive Sonar Signals,” Aug. 1992-June 1993, $50,000. (Principal Investigator.)
51. The Office of Research & Development (Non-Military Government Agency), Time-Frequency Quadratic Processing, Aug. 92 - Aug. 93, $50,000. (Principal Investigator.)
52. Boeing Commercial Airplane Company, “Advanced Signal Processing and Classification for Manufacturing,” Jan. 1993 - Dec. 1993. $120,000 plus $65,000 supplement awarded in June of 1993. (Principal Investigator.)
53. Washington Technology Center, “Advanced Signal Processing for Manufacturing,” July, 1993 - June, 1995, $50,000. (Principal Investigator.)
54. Boeing Commercial Airplane Company, “Representation and Classification of Manufacturing Sensor Signals,” Jan. 1994 - Dec. 1994. $85,000 plus supplement of $80,000 awarded in Dec. of 1994 and extended to Sept. 15, 1995. (Principal Investigator.)
55. Boeing Commercial Airplane Company, “Representation and Classification of Manufacturing Sensor Signals,” Jan. 1995 - Dec. 1995. $83,485. (Principal Investigator.)
56. Daniel H. Wagner Associates, CA, “Automatic Speaker Verification,” (SBIR Phase II from the Army), June - Dec. 1996. $73,188. (Principal Investigator.)
57. Boeing Commercial Airplane Company, “Representation and Classification of Manufacturing Sensor Signals,” Jan. 1996 - Dec. 1998, $228,944. (Principal Investigator.)
58. Virtual DSP Corporation and Washington Technology Center, “Low Latency, High Accuracy Musical Pitch Recognition,” July 1996 - June 1997, $33,000. (Principal Investigator).
59. Office of Naval Research, “Discrete Operator Techniques for Time Frequency Signal Representations,” October 1996 - September 1999. $138,456. (Principal Investigator.)

Patents:

1. L. Atlas, R. Marks II and S. Oh, Optical Neural Network Memory, U.S. Patent No. 4,849,940, July 18, 1989.
2. L. Atlas and M. Vinton, Scalable and Perceptually Ranked Signal Coding, U.S. Patent No. 7136418, Nov. 28, 2002.
3. L. Atlas and J. Thompson, Single Channel Sound Separation, US Patent No. 10/406,802, April 2, 2003.
4. L. Atlas and P. Clark, Gap Interpolation in Acoustic Signals Using Coherent Demodulation, U.S. Patent No. 2009/0048828 A1, Feb. 2009.

5. Kaibao Nie, Les Atlas, Jay Rubinstein, Xing Li, and Pascal Clark, “Enhanced signal processing for cochlear implants”, US Patent No. 8019431 B2, Sep 13, 2011. Licensed by UW CoMotion to Shanghai Listent: www.ee.washington.edu/news/2015/CochlearImplantUsersClosertoHearingMusic.html
6. L. Atlas and Patrick McVittie, “Detecting objects in shipping containers by vibration spectral analysis,” U.S. Patent No. 8,571,829, October 29, 2013.
7. Les Atlas, James Pitton, Thomas Powers, and Scott Wisdom, “Enhancement of Noisy and Reverberant Speech Using Beamforming and Suppression,” U.S. Patent Application 61/990,604 filed May 8, 2014.

Major Service Roles (<u>for the last 10 years</u>)

Electrical Engineering Department:
Chair, Professional Programs (2011-2014)
Electrical Engineering Centennial Chair (2005-2006)
Associate Chair for Research (1998-2001).
Chair, Promotion Review Committees for Prof. Hwang (1996-1998).
Chair, Tenure Review Committee for Prof. Ostendorf (1998-1999).
Chair, Promotion Review Committee for Prof. Dunham (1998-1999).
Chair, Promotion Review Committee for Prof. Riskin (1998-1999).
Chair, Promotion Review Committee for Prof. Bilmes (1999-2000, 2004-2005).
Chair, Promotion Review Committee for Prof. Hui Liu (2000-2001).
Chair, Research Committee (1998-2001).
Chair, Computing Committee (1999-2000).
Chair, Research Strategic Planning Committee (2000-2001).
Chair, Space Committee (2001-2004).
Member, Promotion and Tenure Review Committee (1998-1999).
Member, Department Executive Committee (1998-2001, 2006-present).
Member, Strategic Planning Committee (1999-2000).
Member, Space Committee (1999-2000).
Member, Curriculum Revision Committee (2002-2003).
Member, Awards Committee (2002-2003).

College of Engineering:
Member, Promotion and Tenure Review Committee (2014-2017)
Member, Promotion and Tenure Review Committee (2006-2009)
Chair, Promotion and Tenure Review Committee (2007-2008)
Member, Space Council (2002-2004).
Chair, Fluke Chair Review Committee (1999-2000).
Member, Civil Engineering Chair Search Committee (1997-1998).
Member, EE Department Chair Search Committee (1997-1998).
Member, Osberg Fellowship Committee (1997).

University of Washington:
Elected Member, University of Washington, Graduate Research Council (1996-2000).

National and International Professional Organizations:
- IEEE Signal Processing Society

General Chair of ICASSP ‘98, Seattle, WA, May, 1998.
Chair and Founder, IEEE Signal Processing Society Technical Committee on Signal Processing Theory and Methods, 1998-2000.
Session Chairman for ICASSP, all years 1994-2008.
Chaired a session at the Statistical, Signal and Array Processing Workshop in Portland, OR, 1998
Reviewer for IEEE Trans. Signal Processing (1996-present).
Member, IEEE Signal Processing Society DSP Committee (1996-1998).
Member, IEEE Signal Processing Society Conference Board (1996-1999).
Elected Member-at-Large, IEEE Signal Processing Society Board of Governors (2000-2004).
Member, IEEE Signal Processing Society Technical Committee on Signal Processing Theory and Methods, 2001-2004, 2012-2015.
Organizer, 11th International Workshop on Acoustic Echo and Noise Control, Seattle, Washington, USA, September 14-17, 2008.
Editorial Board, *IEEE Signal Processing Magazine*.

- Funding Agencies and Other Government Service

Ad hoc Advisor, US Office of the Secretary of Defense, 2015-present.
Technology Transfer Thrust Area Leader, ONR Multi-University Research Initiative, 1997-2002.
Co-Chair and Principal Organizer, ONR/NSF Workshop on Signal Processing for Manufacturing and Machine Monitoring (1997).
Review panels for NSF CISE (1997-present).
Research Reviews for the Australian Commonwealth Scientific and Industrial Research Organisation (1996-2006)

Abridged Curriculum Vitæ

Les E. Atlas, Ph.D.

Department of Electrical Engineering, University of Washington
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Research Interests:
Signal processing for dynamical systems, time-frequency analysis, nonstationary statistics, with applications in streaming data, acoustics, machine monitoring, sensor arrays, speech, and auditory sciences. Past impact has included the first publication of trained (linear) convolutional neural networks for temporal signals, now widely used within deep nets. Also, his publication, "Improving generalization with active learning," initiated the machine learning areas of active learning and selective sampling. Graduate students mentored by Prof. Atlas went on to initiate machine learning at Google (David Cohn, now retired), to receive multiple Technical Emmy Awards (Mark Vinton, Dolby Labs), and to top academic and research laboratory positions.

Professional Positions:
Professor of Electrical Engineering and Adjunct Professor of Computer Science and Engineering, University of Washington, Sept. 1994 - Sept. 1998, Sept. 2001 - present.
Visiting Scholar, Centre for the Neural Basis of Hearing, Cambridge University, Cambridge, UK, Feb. 2004 - June 2004.
Visiting Senior Scientist, Fraunhofer Institute IIS, Erlangen, Germany, Aug. 2003 – Jan. 2004.
Associate Chair for Research, Professor of Electrical Engineering, and Adjunct Professor of Computer Science and Engineering, University of Washington, Sept. 1998 – Sept. 2001.
Distinguished Visitor, Queensland University of Technology, Department of Electrical Engineering, Queensland, Australia, January 1994 – Sept. 1994.
Associate Professor of Electrical Engineering and Adjunct Associate Professor of Computer Science and Engineering, University of Washington, Sept. 1988 - Sept. 1994.
Assistant Professor of Electrical Engineering, University of Washington, Jan. 1984 - Sept. 1988.
Part-time member of technical staff, SRI International, Feb. 1982 - Nov. 1983.
Research Assistant, Stanford University, July 1978 - Dec. 1983.

Consulting: Behringer Audio, Cantametrix (acquired by GraceNote and then Microsoft), Erudite, Fraunhofer IIS, The Mitre Corporation, Nintendo, Quest Integrity Group, Silicon Speech, and others.

Education:

University	Dates	Degree
Stanford	9/78 - 12/83	Ph.D. (in Elec. Eng.)
Stanford	9/77 - 9/78	M.S. (in Elec. Eng.)
University of Wisconsin	9/73 - 5/77	B.S.E.E. (high honors)

Other Key Experience and Professional Memberships:
Chair, Jelinek Memorial Workshop on Machine Learning for Speech and Language Processing, 1992.
Co-Chair, AFOSR/ARO EARs Auditory Signal Representation Workshop, 2011.
Member, IEEE Signal Processing Society Tech. Comm. on Speech and Language Processing, 2010-

Member, Editorial Board, *IEEE Signal Processing Magazine*, 2008-13.
Co-Chair, 11th International Workshop on Acoustic Echo and Noise Control, Seattle, WA, 2008.
Chair, University of Washington College of Engineering Council on Promotion and Tenure, 2007-8.
Member-at-Large, Board of Governors, IEEE Signal Processing Society, 2000-6.
Founding Chair, IEEE Signal Processing Society Tech. Comm. on Theory and Methods, 1998-2004.
General Chairman, IEEE Int. Conference on Acoustics, Speech, and Signal Processing (ICASSP '98).
Co-Chair, ONR/NSF Workshop on Signal Processing for Manufacturing and Machine Monitoring, 1997.
General Chairman, IEEE International Symposium on Time-Frequency and Time-Scale Analysis, 1992.

Honors and Awards:

- Bloedel Scholar Award, W. Merrill Bloedel Hearing and Speech Institute, 2012-15
- Chair's Award, University of Washington Electrical Engineering, 2011.
- Nominated as IEEE Distinguished Lecturer by the Signal Processing Theory and Methods Committee of the IEEE Signal Processing Society, Nov. 2011.
- Presented an invited tutorial on Modulation Frequency and Superposition for DSP at the large (1800 attendees) IEEE ICASSP 2008 Conference, Las Vegas, NV, April 30, 2008.
- Presented an invited tutorial on Modulation Frequency for speech and audio at the large (1200 attendees) Interspeech 2007 Conference, Antwerp, Belgium, August 27, 2007.
- Keynote Speaker on Active Learning, West Coast Port Security Workshop, January 13, 2007.
- Fulbright Senior Scholar (with study in Germany), 2003-4.
- Fellow of the IEEE, "For contributions to time-varying spectral analysis and acoustical signal processing," 2004.
- Teacher-of-the-Year, University of Washington Electrical Engineering 2003.
- Department Service Award, University of Washington Electrical Engineering 2003.
- Plenary Speaker, Speech Dynamics by Ear, Eye, Mouth And Machine, An Interdisciplinary Workshop Organized by The Institute for Electronics, Information and Communication Engineers (IEICE) and The Acoustical Society of Japan., Kyoto, Japan, June 27, 2003.
- Keynote Speaker, Conf. On Advanced Signal Signal Processing Algorithms, Architectures, and Implementations, Int. Symp. On Optical Science and Technology, July, 2001, San Diego.
- One of ten experts' summary speakers, IEEE International Conference on Acoustics, Speech, and Signal Processing, June 2000, Istanbul, Turkey.
- Invited as Plenary Speaker, Symposium 2000 on Adaptive Systems for Signal Processing, Communications and Control, Alberta, Canada, October 2000.
- One of ten experts' summary speakers, IEEE International Conference on Acoustics, Speech, and Signal Processing, March 1999, Phoenix.
- Annual Top SAE Paper Award, 1997: McLaughlin, J., L. Atlas, L. Owsley, and G.D. Bernard, "Advances in Real-Time Monitoring of Acoustic Emissions," *Proceedings of the 1997 SAE Aerospace Manufacturing Technology Conference*, Seattle, WA, June, 1997.
- Keynote Speaker, International Symposium - Anthropomorphic Systems of Automatic Speech Recognition and Synthesis, St. Petersburg, Russia, June 30-July 2, 1993.
- Nominated as IEEE Distinguished Lecturer by the Neural Network Committee of the IEEE Signal Processing Society, Nov. 1992.
- Keynote Speaker, Phillips Doppler Ultrasound Conference, June 12, 1991.
- Chosen as one of the 100 Seattle residents under 40 who will be "...shaping the 1990's." (*Seattle Weekly*, July 4, 1989).
- University of Washington, College of Engineering Nominee for NSF Waterman Prize, 1987.
- National Science Foundation Presidential Young Investigator Award, 1985 - 1989.
- Physio-Control Career Development Award, 1984-87.

Publications:

Top Cited and Wide-Impact Papers

1. D Cohn, L Atlas, and R Ladner, "Improving generalization with active learning," *Machine learning* **15** (2), pp. 201-221, 1994. (Started the field of active learning. Called "the CAL Algorithm.")
2. DC Park, MA El-Sharkawi, RJ Marks, LE Atlas, and MJ Damborg, "Electric load forecasting using an artificial neural network," *IEEE Transactions on Power Systems,* **6** (2), 442-449, 1991. (Since showing neural network performance that can exceed human experts, this approach became the standard used in the power industry for electric load forecasting.)
3. JT Connor, RD Martin, and LE Atlas, "Recurrent Neural Networks and Robust Time Series Prediction," *IEEE Transactions on Neural Networks,* **5** (2), pp. 240-254, 1994.
4. J Y Zhao, LE Atlas, RJ Marks, "The use of cone-shaped kernels for generalized time-frequency representations of nonstationary signals," *IEEE Transactions on Acoustics, Speech and Signal Processing*, **38** (7), pp. 1084-1091, 1990.
5. T Homma, L Atlas, and R Marks, "An Artificial Neural Network for Spatio-Temporal Bipolar Patters: Application to Phoneme Classification," *Advances in Neural Information Processing Systems* **1**: 31–40, 1988. (Cited as the first demonstration of convolutional, shift-invariant neural networks)

Other Related Papers in Medical Science, Modeling, and Signal Processing at Low Signal-to-Noise Ratios (58 other previous journal papers and over 200 reviewed conference papers are not listed)

6. Xing Li, K. Nie, N. Imennov, J.H. Won, W. Drennan, J. Rubinstein, & L. Atlas, "Improved perception of speech in noise and Mandarin tones with acoustic simulations of harmonic coding for cochlear implants," *Journal of the Acoustical Society of America*, vol. 132, no. 5, pp. 3387-3398, Nov. 2012.
7. P. Clark, I.P. Kirsteins, and L. Atlas, "Existence and estimation of impropriety in real rhythmic signals," *Proc. IEEE ICASSP*, Kyoto, Japan, 2012.
8. E. C. Larson, J. Froehlich, T. Campbell, C. Haggerty, L. Atlas, J. Fogarty, and S. N. Patel, "Disaggregated Water Sensing from a Single, Pressure-based Sensor: An Extended Analysis of HydroSense using Staged Experiments" *The Pervasive and Mobile Computing (PMC) Journal*, San Francisco, California, June 12-15, 2011.
9. B. King and L. Atlas, "Single-Channel Source Separation Using Complex Matrix Factorization," *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, vol. 19, no. 8, pp. 2591-2597, Nov. 2011.
10. Zweig et al., "Speech Recognition with Segmental Conditional Random Fields: A Summary of the JHU CLSP 2010 Summer Workshop," *Proc. IEEE ICASSP*, Prague, Czech Republic, May 22-27, 2011.
11. P. Clark, I. Kirsteins and L. Atlas, "Multiband analysis for colored amplitude-modulated ship noise," *Proc. IEEE ICASSP*, Dallas, TX, 2010.
12. S.D. Hawley, L.E. Atlas, and H.J. Chizeck, "Some Properties of an Empirical Mode Type Signal Decomposition Algorithm," *IEEE Signal Processing Letters*, vol. 17, 2010, pp. 24-27.
13. P. Clark and L. Atlas, "Time-frequency coherent modulation filtering of non-stationary signals," *IEEE Trans. Signal Process.*, vol. 57, no. 11, pp. 4323-4332, November 2009.
14. J-H. Won, Steven M. Schimmel, Ward R. Drennan, Pamela E. Souza, Les Atlas, Jay T. Rubinstein, "Improving performance in noise for hearing aids and cochlear implants using coherent modulation filtering," *Hearing Research*, May 2008, pp. 1-11. (This paper was the issue's featured cover article)
15. S. Philips, L. Atlas, and J. Pitton, "Auralization of impulsive-source active sonar echoes for perceptual feature identification," *U.S. Navy J. Underwater Acoust.*, 45-62, 2007.
16. Sukittanon, S. Atlas, L., Pitton, J., and Filali, K., "Enhanced modulation spectrum using space-time averaging for in-building acoustic signature identification," *Proc. IEEE ICASSP '06*, Toulouse, France, May, 2006.

17. Schimmel, S., Fitz, K., and Atlas, L., "Frequency reassignment for coherent modulation filtering," *Proc. IEEE ICASSP '06*, Toulouse, France, May, 2006.
18. Bassingthwaighte JB, Chizeck HJ, Atlas LE, and Qian H., "Multiscale modeling of cardiac cellular interactions" pp. 395-424 in **The Communicative Cardiac Cell**, Ann. New York Acad. Sci. 1047 (Sideman S, Beyar R, and Landesberg A., eds.), 2005.
19. Takahiro Shinozaki, Mari Ostendorf, and Les Atlas, "Characteristics of Speaking Style and Implications for Speech Recognition," *J. Acoust. Soc. Of Am.*, 2009 Sep;126(3):1500-10.
20. P. Clark and L. Atlas, "Time-frequency coherent modulation filtering of non-stationary signals," *IEEE Trans. Signal Process.*, vol. 57, no. 11, pp. 4323-4332, November 2009.
21. LE Atlas, DA Cohn, and RE Ladner, "Training connectionist networks with queries and selective sampling," *Advances in Neural Information Processing Systems 2* (NIPS 1989), pp. 566-573.

Principal or Co-Principal Advisor to 28 Ph.D. Students. Most notable graduates:
Faculty
Yunxin Zhao, Professor, Computer Science Department, University of Missouri, Columbia, MO. Advisor: Les Atlas.
Patrick Loughlin, Professor, Dept. of Bioengineering, University of Pittsburgh, Pittsburgh, PA. Advisor: Les Atlas.
Somsak Sukittanon, (Audio) Professor, Department of Engineering, University of Tennessee, Martin, TN. Advisor: Les Atlas.
Eric Larson, Assistant Professor, Department of Computer Science and Engineering, Southern Methodist University, Dallas, TX. Co-Advisors, Les Atlas and Shwetak Patel.
James Pitton, (Acoustics and sonar) Affiliate Associate Professor, Electrical Engineering and Senior Principal Engineer, Applied Physics Lab, University of Washington. Advisor: Les Atlas.

Industry
Dr. Brad Gillespie (Dissertation on modeling and reduction of reverberation), Lead Partner, IA Ventures, New York, NY. Advisor: Les Atlas.
Dr. Jerome Connor (Dissertation with pioneering results for recurrent neural nets), Director, UniCredit Marketing & Investment Banking, London, UK. Advisor: Les Atlas.
Dr. David Cohn (Dissertation initiated the field of active learning), at Google, Mountain View. (His lectures helped initiate machine learning. Active learning improved the efficacy of AdSense.) Co-Advisors, Les Atlas and Richard Ladner.
Roy Peterson (Thesis on DSP for ultrasound), Director, Clinical Science Radiology Ultrasound, Phillips Healthcare, Bothell, WA. Advisor: Les Atlas.
Dr. Siva Bala Narayanan, Chief Executive Officer, Minmini Corporation (Software for cloud and embedded systems), Seattle, WA. Advisor: Les Atlas.
Michael Dougherty (Thesis on sperm whale ID via acoustic tracking), Past leadership: Chief Executive Officer, Quorus, Co-Founder, Redfin, Seattle, WA. Advisor: Les Atlas.
Jeff Thompson, Director R&D, DTS, Inc., (Emphasis on 3-D and high-quality audio) Kirkland, WA. Advisor: Les Atlas.
Cameron Colpitts, Director, RealSelf, (Simulations which show effects of cosmetic procedures) Seattle, WA. Advisor: Les Atlas.
Mark Vinton (Thesis on new approach to high-quality audio coding), Dolby Labs, Won 2 Technical Emmy Awards for Outstanding Technical Achievement. Advisor: Les Atlas.
Dr. Jasha Droppo (Dissertation on audio features for speech recognition), Microsoft Research (speech and audio). Advisor: Les Atlas.
Dr. John Keane. Fred Hutchinson. Advisor: Les Atlas.
Dr. Steven Schimmel, Google Research (speech and audio). Advisor: Les Atlas.
Dr. Charles Pascal Clark, Apple (speech and audio). Advisor: Les Atlas.
Dr. Lane Owsley, UW Applied Physics Lab. Advisor: Les Atlas.

Dr. Jack McLaughlin, UW Applied Physics Lab. Advisor: Les Atlas.
Brandon Smith, DTS (3-D and high-quality audio), Kirkland, WA. Advisor: Les Atlas.
Travis Wilkins, Intel, OR. Advisor: Les Atlas.
Laura Johnston, Hewlett Packard. Advisor: Les Atlas.
Mark Donnell, Applied Physics Lab (Acoustics), Austin, TX. Advisor: Les Atlas.
Dr. Brian King (Dissertation on sound source separation), Amazon Echo (Audio and speech), Seattle. Advisor: Les Atlas.

Recent Research Grants and Contracts (including earlier ONR MURI):

1. Office of Naval Research, Complementary Statistics and Machine Learning for Underwater Acoustics, October, 2013 – June 2017. $341,354. (Co-Principal Investigator). Teamed with PI James Pitton at UW-Applied Physics Lab.
2. Wallace H. Coulter Foundation, Translational Research for Tonality in Cochlear Implants, July 2014 - June 2016, $220,000 direct costs. (Principal Investigator).
3. Jelinek Summer Workshop on Speech and Language Technology, Summer 2015, $574,937 direct costs. (Principal Investigator).
4. US Government Intelligence Community, funding for Postdoctoral Fellow in audio analysis, October 2012- Sept 2014, $220,000. (Principal Investigator).
5. Army Research Office, "Complementary Correlation: New Features for Audio Analysis and Enhancement," September 2012 – December 2016, $349,000. (Principal Investigator).
6. Adobe, Incorporated, "Gift for Modulation Audio Filtering Research," $50,000 direct costs. (Principal Investigator).
7. Office of Naval Research, "Complex Modulation and Passive Sonar," October, 2007 – December 2012. $641,354. (Principal Investigator). Teamed with Dr. Ivars Kirsteins of NUWC-Newport.
8. Air Force Office of Scientific Research, "Sparseness and Demodulation Signal Processing," October 2008 – December 2012. $920,241. (Principal Investigator)
9. Boeing, In Situ Acoustic and Vibration Monitoring for Drilling in Composites using Demodulation Signal Processing. (Co-PI, with M. Ostendorf, PI.) About $95K/year for Atlas. Joint with Prof. M Ramulu, UW Mechanical Engineering. 2010-12.
10. UW CSNE Center, Summer 2012-Spring 2014, 2 years PDRA Support for Elliott Saba, $94,292 direct costs. (Principal Investigator, with Prof. K.C. Lee, UW I-Labs)
11. Multi-University Research Initiative administered by the Office of Naval Research, "Center for Acoustics and Auditory Research," Apr. 1997 – Apr. 2003, $3,900,000 total (Joint with U. Maryland and Boston U. Atlas was UW Principal Investigator and MURI Director of Technology Transfer.).

Patents:

1. L. Atlas, R. Marks II and S. Oh, Optical Neural Network Memory, U.S. Patent No. 4,849,940, July 18, 1989.
2. L. Atlas and M. Vinton, Scalable and Perceptually Ranked Signal Coding, U.S. Patent No. 7136418, Nov. 28, 2002.
3. L. Atlas and J. Thompson, Single Channel Sound Separation, US Patent No. 10/406,802, April 2, 2003.
4. L. Atlas and P. Clark, Gap Interpolation in Acoustic Signals Using Coherent Demodulation, U.S. Patent No. 2009/0048828 A1, Feb. 2009.
5. Kaibao Nie, Les Atlas, Jay Rubinstein, Xing Li, and Pascal Clark, "Enhanced signal processing for cochlear implants", US Patent No. 8019431 B2, Sep 13, 2011. Licensed to Shanghai Listengate.
6. L. Atlas and Patrick McVittie, "Detecting objects in shipping containers by vibration spectral analysis," U.S. Patent No. 8,571,829, October 29, 2013.
7. Les Atlas, James Pitton, Thomas Powers, and Scott Wisdom, "Enhancement of Noisy and Reverberant Speech Using Beamforming and Suppression," U.S. Patent Application 61/990,604 filed May 8, 2014.