UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

SONY ELECTRONICS INC.,

Petitioner,

v.

JAWBONE INNOVATIONS, LLC,

Patent Owner.

Patent No. 8,503,691
Filing Date: June 13, 2008
Issue Date: August 6, 2013

Inventor: Gregory C. Burnett
Title: VIRTUAL MICROPHONE ARRAYS USING DUAL OMNIDIRECTIONAL MICROPHONE ARRAY (DOMA)

DECLARATION OF AKBAR M. SAYEED, Ph.D.

Case No. IPR2023-01228

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I, Akbar M. Sayeed, declare as follows:

1. I have been asked by counsel for Patent Owner, Jawbone Innovations, LLC ("Jawbone" or "Patent Owner"), to review U.S. Patent No. 8,503,691 (the "'691 Patent") entitled VIRTUAL MICROPHONE ARRAYS USING DUAL OMNIDIRECTIONAL MICROPHONE ARRAY (DOMA), and to provide my technical review, analysis, insights, and opinions regarding the '691 Patent in view of the prior art cited by Petitioner Sony Electronics Inc. ("Sony" or "Petitioner"). I submit this declaration in support of Patent Owner's Preliminary Response in this IPR proceeding. I have personal knowledge of the matters stated herein and would be competent to testify to them if required.

2. I have been retained on behalf of Jawbone for the above-captioned *inter partes* review proceeding. I understand that the '691 Patent is currently assigned to Jawbone.

3. I am over 18 years of age. I have personal knowledge of the facts stated in this Declaration and could testify competently if asked to do so.

I. INTRODUCTION

A. Background and Qualifications

4. I have reviewed and am familiar with the specification of the '691 Patent. I understand the '691 Patent has been provided as Exhibit 1001. I will cite to the specification using the following format: '691 Patent at col.:line.

5. My CV is being submitted simultaneously herewith (Ex. 2002).

6. I received my B.S. degree in Electrical and Computer Engineering (ECE) from the University of Wisconsin-Madison in 1991. I received my M.S. and Ph.D. degrees in ECE from the University of Illinois at Urbana-Champaign in 1993 and 1996, respectively. In my undergraduate and graduate studies, I look a variety of courses generally focusing on the areas of signal processing, communication theory, information theory, electromagnetics and antennas, and statistical techniques in signal processing and communications. My Ph.D. dissertation was entitled "Statistical Time-Frequency Analysis" in which I proposed a new framework for statistical signal processing using time-frequency representations, mathematical tools that extend the powerful theory of Fourier transforms and Fourier analysis to time-varying signals and systems.

7. I currently work as an Independent Researcher, Engineer and Technical Consultant, and worked as professor of Electrical and Computer Engineering at the University of Wisconsin-Madison from 1997-2021, where I directed the *Wireless Communications and Sensing Laboratory* until my retirement on August 1, 2021 to pursue a career as an independent researcher and consultant.

8. My current work as an independent researcher and consultant spans STEM (science, technology, engineering & mathematics) fields through the lens of

information science and technology. My STEM skills and experience cover a broad range of areas, including:

- Wireless technologies for sensing, processing and communication of information (5G/6G/XG), internet of things, and emerging technologies for untapped spectrum (e.g., millimeter-wave, THz).
- Sensing and acquisition of data in new modalities through technological innovation.
- Machine learning and statistical techniques for extracting useful information from data.
- Quantum information science and technology (computing, sensing and communication) spanning foundational concepts, algorithms, and platforms.
- Conception, design and development of new information technologies through basic theory, computational modeling, and hybrid software-hardware prototyping and experimentation.

9. I have written/co-written 200+ papers in leading journals and conferences. I am an inventor/co-inventor of 10 patents.

10. After receiving my Ph.D. in 1996, I spent a year at Rice University as a postdoctoral research fellow where I expanded the scope of my research to the growing field of wireless communications that underpins much of mobile cellular

technology. In particular, the propagation channels encountered in mobile cellular communication are inherently time-varying in nature and that provided a natural connection to my Ph.D. research. After spending a year at Rice University, I accepted a position as an Assistant Professor in the ECE Department at the University of Wisconsin-Madison in 1997. I was granted tenure and promoted to Associate Professor in 2003, and was promoted to Professor in 2008.

11. I founded and directed the Wireless Communication and Sensing Laboratory at UW-Madison and engaged in research, teaching and innovation in the basic science, engineering, and technology of sensing, processing, and communication of information. In particular, I led a team in the conception, design, development and successful demonstration of a state-of-the-art prototype for multi-beam MIMO (multiple input multiple output) communication at 28 GHz based on the concept of a lens array that I had pioneered.

12. I co-founded and co-led the NSF-sponsored Research Coordination Network (RCN) on millimeter-wave wireless (2016-2019) for bringing together researchers from academia, industry, and government agencies to address critical technical challenges in the area.

13. From 2017-2019 I served as a Program director in the Electrical, Communications, and Cyber Systems (ECCS) Division of the Directorate of Engineering at the National Science Foundation. At the NSF, I managed existing

programs and initiated new research programs involving sensing, processing, and communication of information in all possible physical modalities, architectures, and technologies, including quantum information science and engineering. I actively contributed to three NSF-wide working groups: Electromagnetic Spectrum Management (ESM) working group, and the working groups of two of the 10 NSF Big Ideas: i) Quantum Leap and ii) Harnessing the Data Revolution. In particular, I co-led the team that developed the NSF solicitation for Quantum Leap Challenge Institutes (Feb 2019) in response to the National Quantum Initiative (NQI) Act signed in December 2018.

14. I have a demonstrated history (25 yrs.) of leading an interdisciplinary research and development group at the forefront of foundational and technological innovations in wireless communication and sensing. Physics-based accurate and computationally tractable system modeling has been a key underlying theme in my work, spanning communication & sensor networks, multipath propagation over highly dynamic environments, wideband MIMO systems, mm-wave and THz wireless, and prototype development (hardware & software).

15. My technical background, experience and skills are directly relevant to the subject matter in this case. The modeling, analysis and processing of information bearing signals and waves, including transmission and reception through antennas and antenna arrays, and propagation over channels, has been an underlying theme

throughout my career. In particular, I have extensive experience in the conception, modeling, development and analysis of techniques and algorithms for processing and filtering of signals from an array of sensors or antennas, including beamforming, interference and noise suppression, and adaptive filtering methods. While most of my work has focused on electromagnetic signals, the underlying physics and mathematics are very similar to acoustic signals.

II. COMPENSATION

16. My compensation for time worked on this proceeding is not dependent on any issues related to the ’691 Patent, the outcome of this proceeding, or the substance of my opinions. My compensation for time worked on this proceeding is at my customary rate of $200 per hour. I have no financial interest in, or affiliation with, the Patent Owner or any of the real parties in interest.

III. MATERIALS CONSIDERED

17. In providing my technical review, analysis, insights, and opinions, I have considered the ’691 Patent and its prosecution history.

18. I have also considered the Petition filed by the Petitioner in this proceeding and the relevant exhibits relied on by Petitioner, including the expert declaration submitted by Jeffrey S. Vipperman, Ph.D. (“Ex. 1003”).

19. I have also considered my own experience and knowledge, as discussed above and described more fully in my CV, in areas including signal processing, electrical engineering, and other related technologies.

IV. LEGAL PRINCIPLES

20. I understand that a patent claim is unpatentable as "obvious" if the subject matter of the claim as a whole would have been obvious to a person of ordinary skill in the art (POSITA) as of the time of the invention at issue.

21. I understand that the use of "the person of ordinary skill" rubric is to prevent one from improperly, in the present day, using hindsight to decide whether a claim is obvious.

22. I understand that the following factors must be evaluated to determine whether the claimed subject matter is obvious: (1) the scope and content of the prior art; (2) the difference or differences, if any, between the scope of the patent claim and the scope of the prior art; and (3) the level of ordinary skill in the art at the time of the invention.

23. I understand that certain secondary considerations, such as commercial success, skepticism of experts, surprise, and copying, may provide evidence of non-obviousness. I further understand that such considerations are often the most probative and determinative of obviousness or non-obviousness.

24. I understand that I must construe a claim in accordance with the ordinary and customary meaning of the language of such claim as understood by one of ordinary skill in the art and the prosecution history pertaining to the patent.

V. LEVEL OF SKILL IN THE ART

25. I understand that I should perform my analysis from the viewpoint of a person of ordinary skill in the art. I understand that this hypothetical person of ordinary skill in the art is considered to have the normal skills of a person in a certain technical field. I understand that factors that may be considered in determining the level of ordinary skill in the art include: (1) the education level of the inventor; (2) the types of problems encountered in the art; (3) the prior art solutions to those problems; (4) rapidity with which innovations are made; (5) the sophistication of the technology; and (6) the education level of active workers in the field.

26. In my opinion, a person of ordinary skill in the art would have a bachelor's degree in computer engineering, computer science, electrical engineering, mechanical engineering, or a similar field, and approximately three years of industry or academic experience in a field related to acoustics, speech recognition, speech detection, or signal processing. Work experience can substitute for formal education and additional formal education can substitute for work experience.

VI. THE CLAIMED INVENTION OF THE '691 PATENT

27. I have reviewed the ’691 Patent, entitled VIRTUAL MICROPHONE ARRAYS USING DUAL OMNIDIRECTIONAL MICROPHONE ARRAY (DOMA) (Ex. 1001). It issued on August 6, 2013, and was filed on June 13, 2008.

28. The ’691 Patent discloses and claims dual omnidirectional microphone arrays and devices containing those microphone arrays for use in noise suppression. Unlike “conventional arrays and algorithms which seek to reduce noise by nulling out noise sources,” the ’691 Patent uses omnidirectional microphone arrays “to form two distinct virtual directional microphones which are configured to have very similar noise responses and very dissimilar speech responses.” ’691 Patent, Abstract. While “conventional multi-microphone systems attempt to increase the [signal-to-noise ratio] of the user’s speech by ‘steering’ the nulls of the system to the strongest noise sources, [t]his approach is limited in the number of noise sources removed by the number of available nulls.” *Id.*, 1:46-51. Counterintuitively, in the ’691 Patent, “[t]he only null formed by the DOMA is one used to remove the *speech* of the user from [the second virtual microphone]”. *Id.* at 4:67-5:2 (emphasis added). However, the similarity of the ’691 Patent’s virtual microphones’ noise responses allows them to be cancelled while leaving the speech response intact, “resulting in excellent noise suppression performance and minimal speech removal and distortion.” *Id.* at 8:1-2; 10:26-42.

29. The '691 Patent discloses a noise suppression apparatus that creates two virtual microphones V_1 and V_2 based on the signals generated by two omnidirectional microphones.

$$V_1(z) = O_1(z)z^{-\gamma} - \beta O_2(z)$$

$$V_2(z) = O_2(z) - z^{-\gamma}\beta O_1(z)$$

Id. at 11:67, 10:56. In these equations, β and γ are defined as follows:

$$\beta = \frac{d_1}{d_2}$$

$$\gamma = \frac{d_2 - d_1}{c} . f_s \text{ (samples)}$$

Id. at 10:59-62. "The distances d_1 and d_2 are the distances from O_1 and O_2 to the speech source," respectively, and "γ is their difference divided by c, the speed of sound, and multiplied by the sampling frequency f_s." *Id.*, 11:1-4.

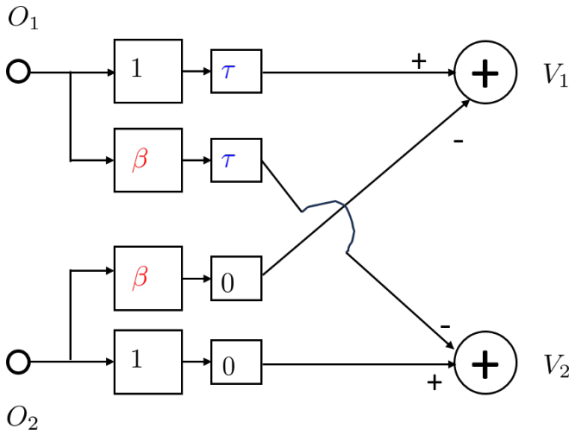
30. In the analog frequency domain, the equations for V_1 and V_2 are written as follows:

$$V_1(f) = e^{-j2\pi f\tau}O_1(f) - \beta O_2(f)$$

$$V_2(f) = -\beta e^{-j2\pi f\tau}O_1(f) + O_2(f)$$

where $z = e^{j2\pi f}$ and $\tau = \frac{d_2 - d_1}{c}$.

31. The configuration of the '691 Patent circuit according to these equations can be shown as follows:



'691 Patent Circuit

In this diagram, the signals from omnidirectional microphones O_1 and O_2 first pass through a gain with a value of β or 1 (i.e., no gain) and then pass through a delay τ or 0 (i.e., no delay). I refer to the four lines of the circuit throughout this declaration as the first, second, third and fourth lines. In this circuit, the first and third lines are summed to form virtual microphone V_1, and the second and fourth lines are summed to form the virtual microphone V_2.

32. The '691 Patent discloses that the apparatus "works very well because the noise (far-field) responses of V_1 and V_2 are very similar while the speech (near-field) responses are very different." *Id.* at 17:5-8.

33. The '691 Patent discloses polar plots depicting similar noise responses which may be cancelled while leaving dissimilar speech responses intact. For instance, Figures 10 and 12 show the linear responses of virtual microphones V_2 and V_1, respectively, to noise from a source at 1 meter:

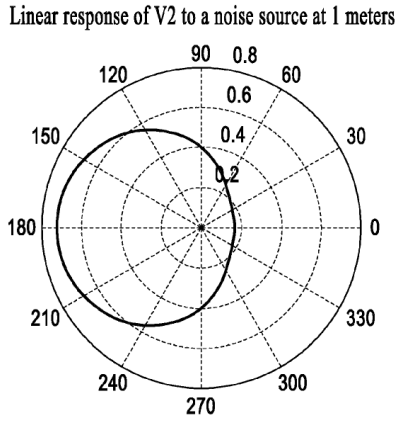


FIG.10

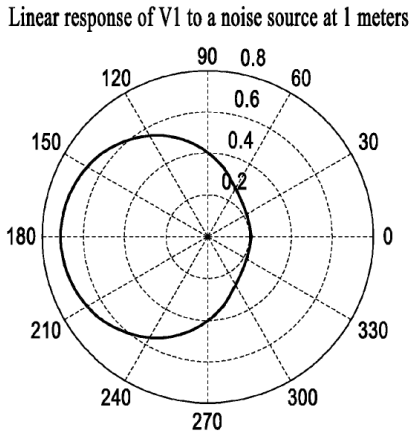


FIG.12

'691 Patent, 11:25-29, 12:25-28, Figs. 10, 12.

34. Figures 9 and 11 show the linear responses of virtual microphones V_2 and V_1, respectively, to speech at a distance of 0.1 meter at an angle of zero degrees:

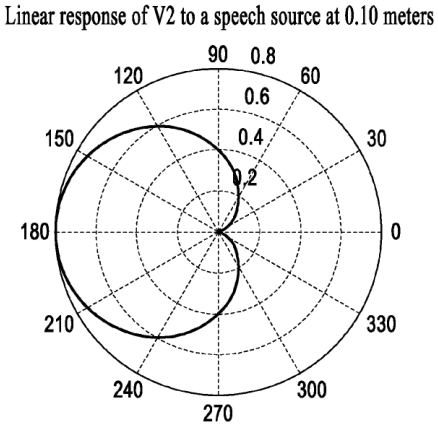


FIG.9

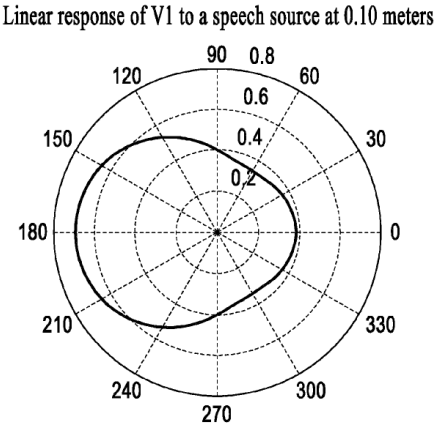


FIG.11

'691 Patent, 11:21-23, 12:20-22, Figs 9,11.

35. Figure 9 shows a null in the linear response of virtual microphone V_2 to speech at zero degrees, "where the speech is typically expected to be located." *Id.*, 11:24-25. As shown in Figure 10, "[t]he linear response of V_2 to noise is devoid of or includes no null, meaning all noise sources are detected." *Id.*, 11:28-29. This configuration ensures that virtual microphone V_2 will detect all of the noise in front

of the user so that it can be removed, which is an advantage over "conventional systems that can have difficulty removing noise in the direction of the mouth of the user." *Id.*, 11:39-40. In addition, "the superior noise suppression made possible using this system more than compensates for the initially poorer SNR," or signal to noise ratio, of the virtual microphones. *Id.*, 12:58-60.

VII. THE ALLEGED PRIOR ART

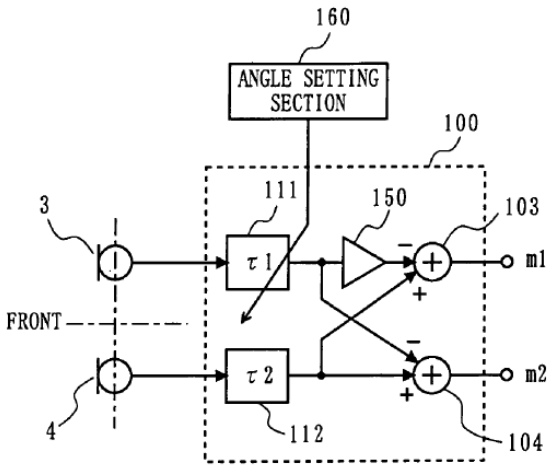
A. Kanamori (U.S. Patent Application Publication No. 2004/0185804)

36. I have reviewed U.S. Patent Application Publication No. 2004/0185804 ("Kanamori") (Ex. 1005). Kanamori was filed on November 18, 2003, and published on September 23, 2004. Kanamori is directed to "a microphone device and an audio player which detects a desired sound coming from a specific direction with noise being suppressed." Ex. 1005, ¶ (0002).

37. Kanamori's system comprises "a microphone device which detects a target sound coming from a direction of the target sound" which includes "a signal generating section, a determining section, an adaptive filter section, a subtracting section, and a noise suppressing section." *Id.*, ¶ (0019). Kanamori's "signal generating section" generates both a "main signal . . . with a sensitivity in the direction of the target sound and a noise reference signal . . . with a sensitivity higher in another direction than in the direction of the target sound." *Id.*

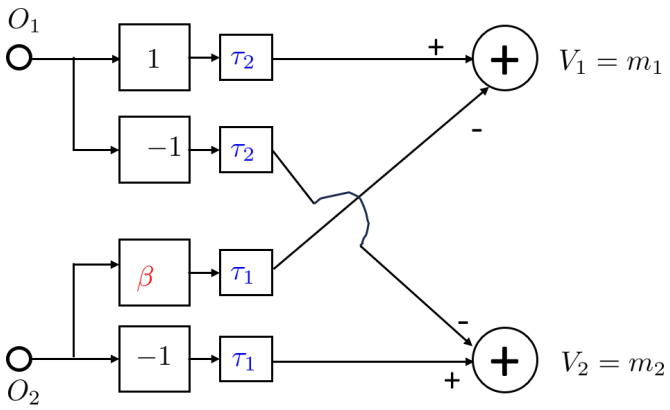
38. Kanamori discusses at least 11 separate embodiments of its system, variously comprising up to six microphones. *Id.*, ¶ (0192). Embodiment 8, on which the Petition relies, is shown below:

FIG. 17A



Ex. 1005, Fig. 17A.

39. The configuration of the Kanamori circuit based on Figure 17A and the supporting disclosure can be shown as follows:



Circuit of Kanamori, Figure 17A

Ex. 1005, Fig. 17A. In this diagram, O_1 corresponds to mic 4 in Figure 17A, and O_2 corresponds to mic 3. Gain β corresponds to signal amplifying section 150 in Figure 17A. The remaining gains of 1 and -1 are provided for consistency with the operation of the signal adders in Figure 17A. The signals pass through a delay τ_1 or τ_2, which correspond to the delays 111 and 112, respectively. The resulting signals are ultimately summed to form m1 and m2.

40. Kanamori discloses that Embodiment 8 as shown in Figure 17A improves on Embodiment 7 by adding an angle setting section 160 and a second signal delaying section 112. Ex. 1005, ¶ (0173). These components permit the target sound direction to be changed in a range of ± 90 degrees with respect to the front of the device. Ex. 1005, ¶ (0175). According to Kanamori, "angle setting section 160 can change a signal delay amount τ1 of the first signal delaying section 111 in a range of $0 \leq \tau1 \leq 2d/c$ (where d is a distance between the microphone units and c is the speed of sound)." Ex. 1005, ¶ (0175). When the second signal delaying section 112 includes a delay amount τ2 set as $\tau2=d/c$, the full adjustment range of ± 90 degrees is achieved. *Id.*

41. Kanamori's system in Embodiment 8, which uses the same microphone arrangement as Embodiment 6, Ex. 1005, ¶¶ (0159), (0162), (0172), assumes that speech and noise are in the far field. Kanamori discloses that the microphones have the same directivity characteristics and are typically required to be a certain distance

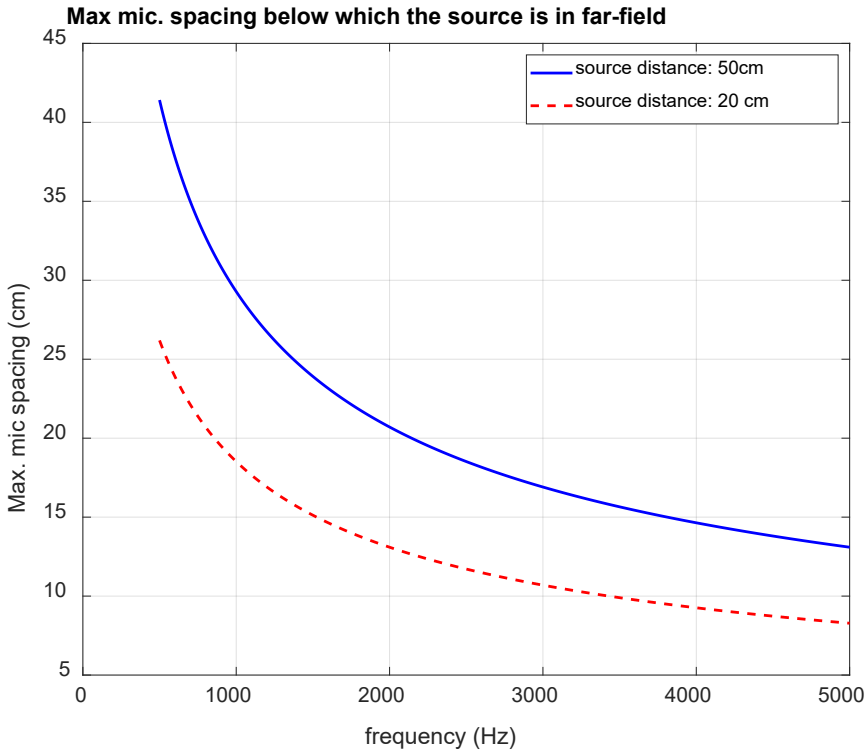
apart, i.e., 1 cm to 5 cm, in a device such as a video recorder. Ex. 1005, ¶¶ (0151), (0159). In the examples shown in Figure 9, the talker is 50 cm from the microphones, and a reflective object is 20 cm from microphones. Ex. 1005, Fig. 9.

42. McCowan states that the near-field cut off for the distance (r) between a source and a microphone (antenna) array is

$$r < \frac{2L^2}{\lambda}$$

where r is the distance from the source to the microphone array, L is the size of the array (the spacing between the two microphones, specifically) and λ is the wavelength of the sound wave which, in turn, is given by λ=c/f, where c is the speed of sound in air (343 m/s in dry air at 20 deg C), and f is the frequency of sound. Ex. 1006 at 89.

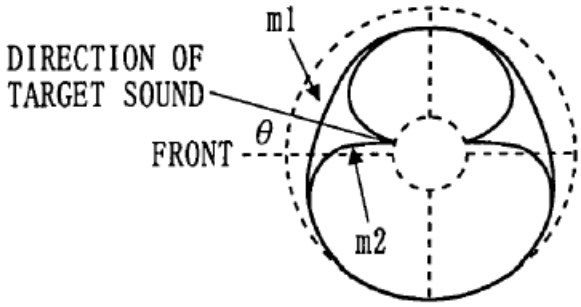
43. Applying the McCowan equation to embodiments of Kanamori where the sound source is 20 cm or 50 cm from the microphones, the sound source would be in the far field even where the microphones are more than 5 cm apart.



As such, Kanamori assumes that speech and the accompanying noise are in the far field.

44. Kanamori's noise reference signal includes signals from both the target sound and noise, requiring an "adaptive filter section" to generate "a signal indicative of a signal component of the target sound included in the noise reference signal." Ex. 1005, ¶ (0019). Consistent with that approach, Kanamori's noise reference signal ("m2") in Embodiment 8 includes two nulls:

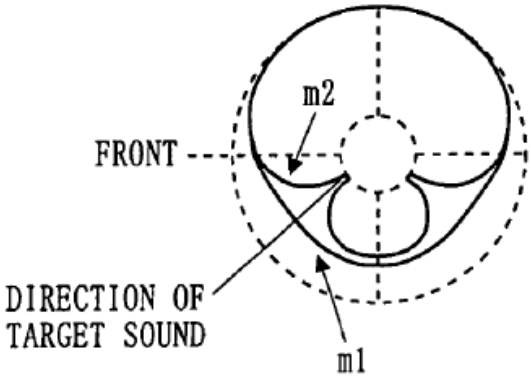
FIG. 17B



Ex. 1005, Fig. 17B

45. Kanamori similarly contemplates a main signal ("m1") with a different directivity pattern than its noise reference ("m2") in a direction away from speech:

FIG. 17C



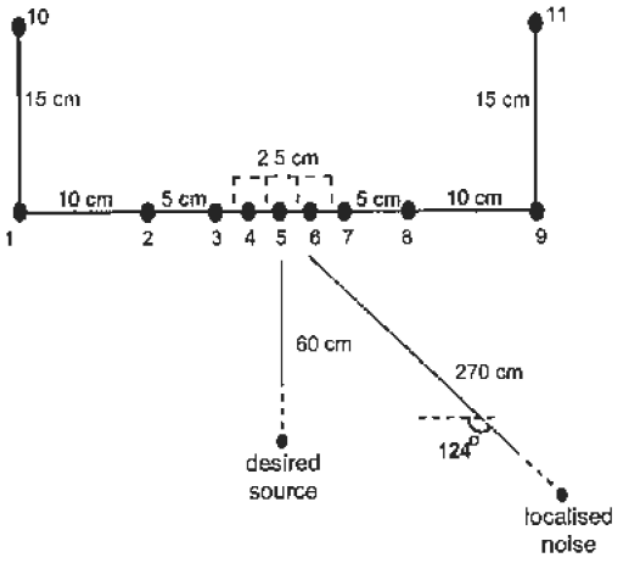
Ex. 1005, Fig. 17C.

46. Kanamori teaches that in "the direction of the target sound, there is a large difference in sensitivity between the signals m1 and m2." Ex. 1005, ¶ (0168). In other directions, "it is preferable that there is no difference insensitivity [sic] therebetween." *Id.*

B. McCowan (Iain A. McCowan et al., *Near-Field Adaptive Beamformer for Robust Speech Recognition*, Digital Signal Processing, Vol. 12, Issue 1 (2002), 87-106)

47. I have reviewed *Near-Field Adaptive Beamformer for Robust Speech Recognition* ("McCowan") (Ex. 1006). McCowan was purportedly published in 2002.

48. McCowan discloses an adaptive noise cancellation system to improve the low frequency directivity of a conventional adaptive beamformer where the speaker is in the microphone array's near field. Ex. 1006 at 87, 104. The microphone array is comprised of 11 microphones where the width of the microphone array is 40 centimeters and its depth is 15 centimeters.



Id. at 94-96, Fig 5. In McCowan's experiment, the microphone array was situated in a computer room with the speaker located at 60 cm from the center microphone and

a noise source located at an angle of 124º and a distance of 270 cm from the array. *Id.* at 96.

49. The microphone array is connected to an adaptive noise cancellation system for near-field speech sources, termed a near-field adaptive beamformer ("hereinafter "NFAB") as shown in Figure 3:

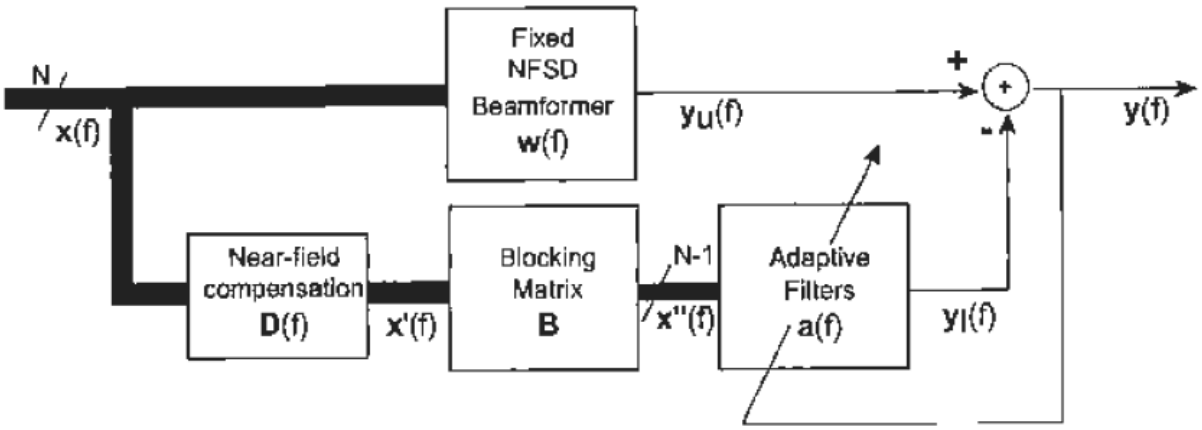


FIG. 3. Near-field adaptive beamformer.

Ex. 1006 at 91, Fig. 3.

50. McCowan's NFAB "is implemented using the standard generalized sidelobe canceler (GSC) system structure, where a near-field superdirective (NFSD) beamformer is used as the fixed upper-path beamformer to improve the low frequency performance." *Id.* at 87. The NFSD is used with "a near-field compensation unit" and "a standard generalized sidelobe canceling blocking matrix and adaptive filters." *Id.* at 90-91.

51. McCowan's NFSD comprises an "upper path," while "[t]he blocking matrix and adaptive filters essentially implement a conventional (nonsuperdirective) beamformer that adaptively focuses on the major sources of noise" comprising a "lower path." *Id.* at 98. The lower path generates a noise estimate which is used to cancel noise in the output signal of the NFSD in the upper path. *Id.*, 87-91, 93-96.

52. As shown in Figs. 6 & 7 below McCowan's upper path NFSD includes four nulls at 300 Hz, while its lower path NFAB includes two nulls at 300 Hz:

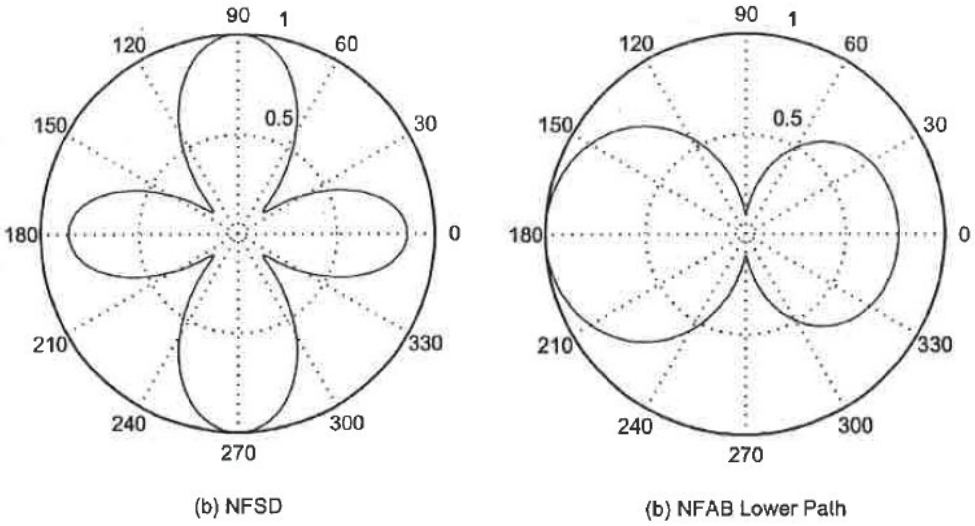


FIG. 6. Upper path directivity pattern at 300 Hz. **FIG. 7.** Lower path directivity pattern at 300 Hz.

Id. at 98-99. The "overall beamformer directivity pattern" of McCowan's NFAB also includes four nulls at 300 Hz. *Id.* at 100.

53. McCowan contemplates that the discussed beamformers will be applied in near-field environments, and specifically notes that it employs "a spherical

propagation model in its formulation, rather than assuming a far-field model." *Id.* at 88.

VIII. CLAIM CONSTRUCTION

54. I understand that Petitioner claims to have applied the plain and ordinary meaning of the claims terms as they would have been understood by a POSITA, consistent with the '691 Patent's disclosure and prosecution history. For my analysis, I have considered each claim term and applied its plain and ordinary meaning consistent with the '691 Patent's disclosure and prosecution history.

IX. PETITIONER'S COMBINATION DOES NOT RENDER ANY CLAIM OBVIOUS

A. GROUND 1: The Combination of Kanamori and McCowan Does Not Render Obvious Claims 1-22 and 29-40

55. In my opinion, the combination of Kanamori and McCowan does not render claims 1-22 and 29-40 of the '691 Patent obvious.

1. The Petition Does Not Identify Legitimate Motivations to Combine the Teachings of Kanamori and McCowan to Arrive at the Claimed Invention

56. In my opinion, the Petition contains no evidence that a POSITA would have been motivated to combine McCowan and Kanamori as set forth in the Petition. The Petition's obviousness analysis is, instead, based on a hindsight reverse engineering of the noise suppression device of the '691 Patent.

i. The Petition Fails to Show that a POSITA Would Have Been Motivated to Implement Kanamori in a Headset

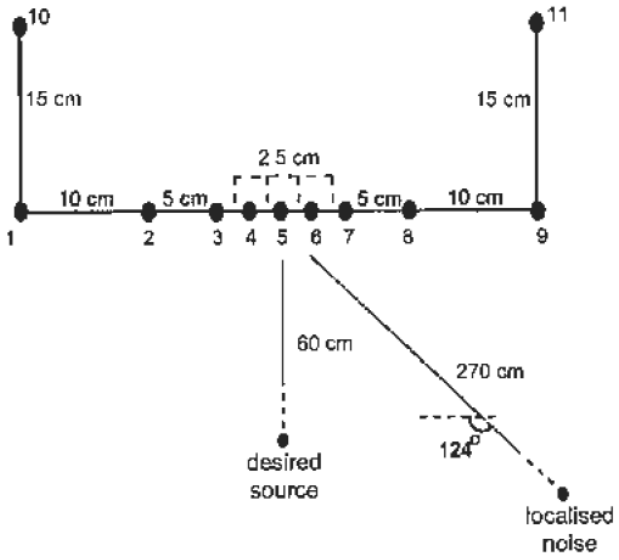
57. The Petition's motivation to modify Kanamori for implementation in a near-field system generally, and a headset specifically, is that a POSITA would recognize that a headset is a near-field implementation of Kanamori's "calling" or "voice recognition" use environments. Ex. 1003, ¶ 69. I disagree that even if a POSITA would recognize that a headset is a "conventional microphone device used for calling," a POSITA would have been motivated to modify Kanamori for near-field application based merely on that recognition. Although the Petition identifies references supposedly disclosing that headsets were known and utilized omnidirectional microphones between 3-5" from a user's mouth, Pet. at 18-19, the Petition does not argue that any of these references would have motivated a POSITA to modify Kanamori to make a headset.

ii. The Petition Fails to Show that a POSITA Would Have Been Motivated to Combine Kanamori with McCowan to Make a Headset

58. In my opinion, even if a POSITA would have been motivated to modify Kanamori to make a headset, the Petition presents no evidence or explanation why a POSITA would have been motivated to combine Kanamori with McCowan to achieve that modification. In particular, the Petition fails to articulate why a POSITA

would have looked to McCowan to improve Kanamori or would have combined McCowan with Kanamori in the specific way described by the Petition.

59. McCowan does not disclose a headset. McCowan describes a microphone array of 11 microphones with a width of 40 centimeters and a depth of 15 centimeters.



Ex. 1006 at 94-96, Fig 5. In my opinion, a POSITA would know that a headset could not incorporate a microphone array which is 40 cm wide and 15 cm deep.

60. McCowan discloses that the purpose of his apparatus is to improve the *low frequency* directivity of a conventional adaptive beamformer where the speaker is in the near field. Ex. 1006 at 87, 104. The Petition does not purport to modify Kanamori to make a headset with improved noise suppression at low frequencies rather than across the entire frequency band for speech.

61. In my opinion, a POSITA would not have been motivated to combine McCowan and Kanamori to arrive at a near-field application such as a headset.

62. The Kanamori embodiment (Embodiment 8) that Petitioner combines with McCowan assumes a target sound and noise in the far field. Kanamori discloses that the microphones are typically required to be a certain distance apart, i.e., 1 cm to 5 cm, in a device such as a video recorder. Ex. 1005, ¶¶ (0151), (0159). In the examples shown in Figure 9, the talker is 50 cm from the microphones, and a reflective object is 20 cm from microphones. Ex. 1005, Fig. 9. In this configuration, the talker is in the far field. *See supra.*, ¶¶ 41-43.

63. The Petition and Dr. Vipperman argue that Kanamori's reference to "calling" in paragraph 207 supplies the motivation to use it in a headset. Pet., 18; Ex. 1003, ¶ 69. But Kanamori, as Petitioner repeatedly admits, is directed only to far-field applications. A POSITA would thus understand the reference to "calling" to refer to a far-field device, such as a table-top speakerphone, and not read Kanamori as encouraging (or even compatible with) implementation into a headset.

64. Indeed, modifying Kanamori as Petitioner suggests would defeat Kanamori's purpose of providing noise cancelation in the far-field. Kanamori's angle setting section steers a null in m2 towards the target sound. Kanamori, [0175],[0176]; Fig. 17B, 17C.

FIG. 17B

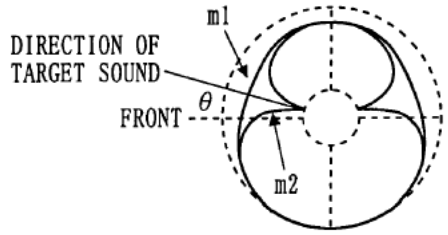
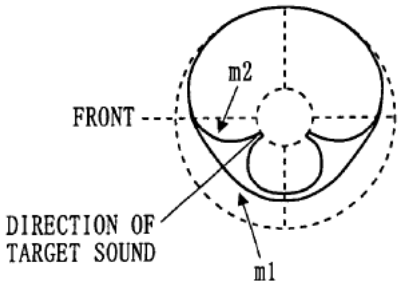
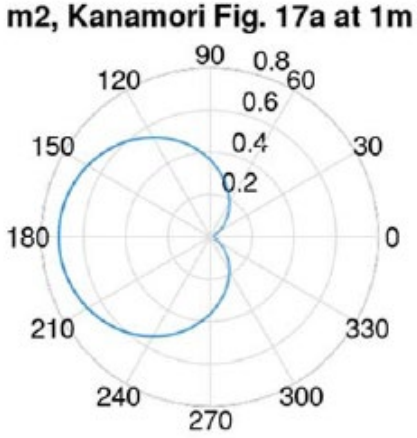
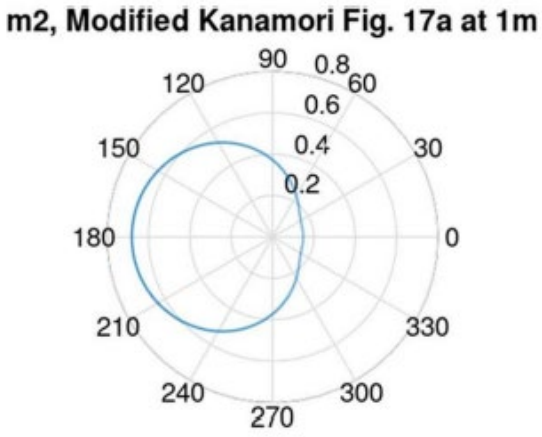


FIG. 17C



65. However, a POSITA would know that implementing Petitioner's proposed combination would remove m2's null in the far-field case. Petitioner's own plots demonstrate the difference:





Pet., 30, 37.

66. A POSITA would therefore know that Petitioner's combination introduces distortion into Kanamori's intended far-field application. A POSITA would not be motivated to do so. The Petition ignores this issue.

67. The Petition therefore does not explain why a POSITA would combine far-field Kanamori with McCowan when the stated motivation to make the combination was to create a ***near-field*** device. McCowan was apparently chosen for combination with Kanamori simply because it discloses a near-field application. Pet. at 28-39. Petitioner looks to McCowan to modify the m2 signal of the Kanamori embodiment to improve denoising of speech in the *near field* even though Kanamori's m2 signal is used for denoising speech in the *far field*. The Petition does not address the inherent inconsistency in these positions.

X. CONCLUSION

68. In signing this declaration, I recognize that the declaration will be filed as evidence in a contested case before the Patent Trial and Appeal Board of the United States Patent and Trademark Office. I also recognize that I may be subject to cross-examination in the case and that cross-examination will take place within the United States. If cross-examination is required of me, I will appear for cross-examination within the United States during the time allotted for cross-examination.

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

Executed this 24th day of October 2023

Respectfully submitted,



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