

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. 10,779,080

Filing Date: July 22, 2013

Issue Date: September 15, 2020

Inventor: Gregory C. Burnett

Title: DUAL OMNIDIRECTIONAL MICROPHONE ARRAY (DOMA)

DECLARATION OF AKBAR M. SAYEED, Ph.D.

Case No. IPR2023-01065

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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. 10,779,080 (the “’080 Patent”) entitled DUAL OMNIDIRECTIONAL MICROPHONE ARRAY (DOMA), and to provide my technical review, analysis, insights, and opinions regarding the ’080 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 ’080 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 ’080 Patent. I understand the ’080 Patent has been provided as Exhibit 1001. I will cite to the specification using the following format: ’080 Patent at col.:line.

5. My CV is being submitted simultaneously herewith (Ex. 2003).

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 took 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 '080 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 '080 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 '080 PATENT

27. I have reviewed the '080 Patent, entitled DUAL OMNIDIRECTIONAL MICROPHONE ARRAY (DOMA) (Ex. 1001). It issued on September 15, 2020, and was filed on July 22, 2013.

28. The '080 Patent discloses and claims methods for noise suppression in dual omnidirectional microphone arrays. Unlike “conventional arrays and algorithms which seek to reduce noise by nulling out noise sources,” the '080 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.” '080 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.*, 1:55-59. Counterintuitively, in the '080 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 3:56-57 (emphasis added). However, the similarity of the '080 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 6:59-62; 9:22-38.

29. The '080 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 9:52, 11:1. In these equations, β and γ are defined as follows:

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

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

Id. at 9:55-57. “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.*, 9:64-67.

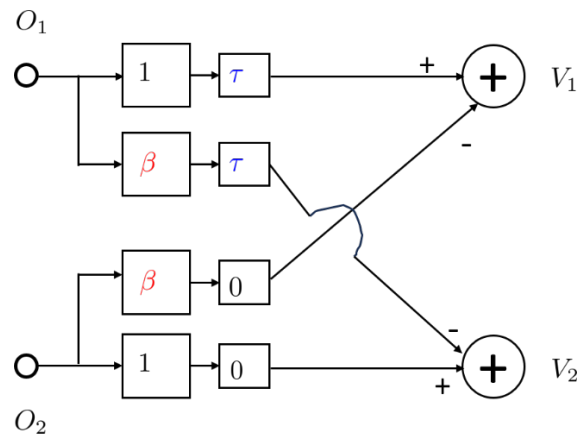
30. In the analog frequency domain, the equations for V_1 and V_2 are written as follows:

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

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

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

31. The configuration of the '080 Patent circuit according to these equations can be shown as follows:

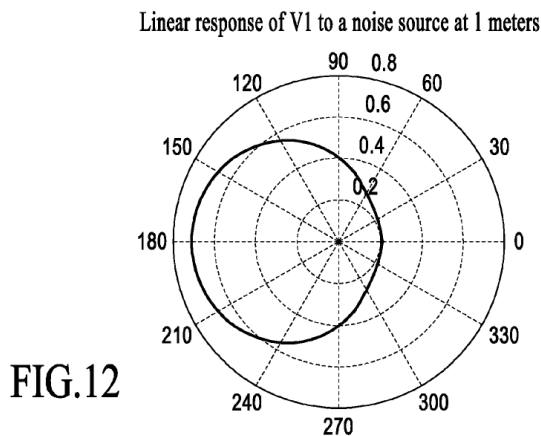
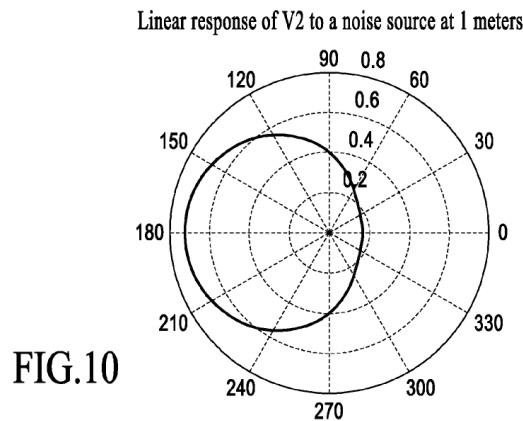


'080 Patent Circuit

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 .

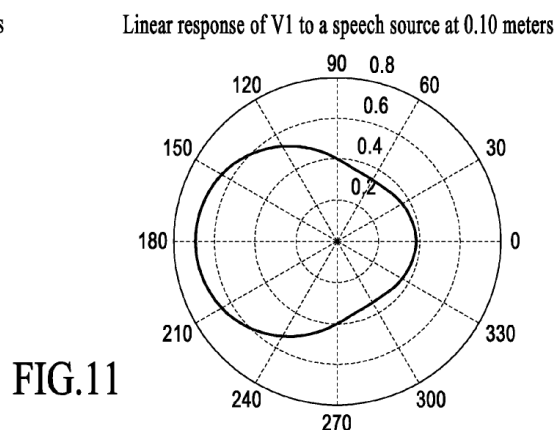
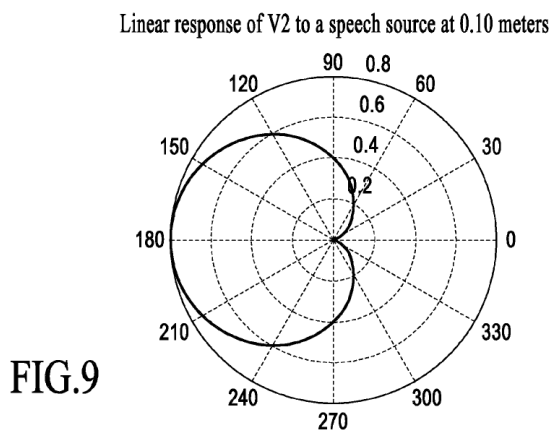
32. The '080 Patent discloses that the apparatus “works very well because the noise (far-field) responses of V_1 and V_2 are very similar while the speech (near-field) responses are very different.” *Id.* at 16:4-6.

33. The '080 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:



'080 Patent, 10:22-25, 11:25-27, 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:



'080 Patent, 10:18-22, 11:24, 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.*, 10:20-22. 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.*, 10:25-27. 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.*, 10:35-38. 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.*, 11:59-61.

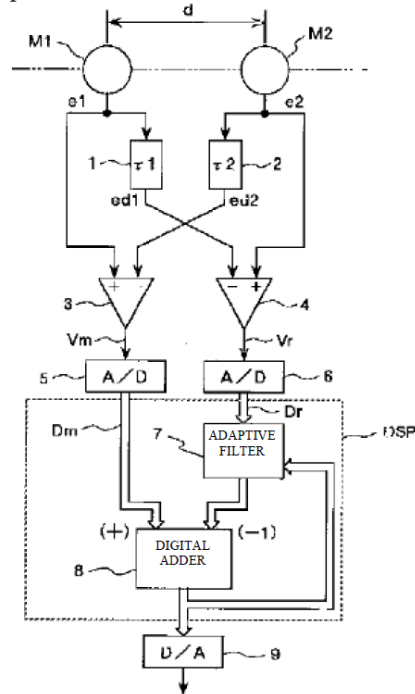
VII. THE ALLEGED PRIOR ART

A. Ikeda (Japanese Unexamined Patent Application No. H11-18186)

36. I have reviewed the certified English translation of Japanese Unexamined Patent Application Publication No. H11-18186 (“Ikeda”) (Ex. 1006). Ikeda published on January 22, 1999.

37. Ikeda discloses an embodiment of a primary pressure gradient microphone with “two omnidirectional microphone units M1 and M2 having essentially the same performance are disposed at a distance d apart from one another.” Ex. 1006, ¶ (0018). The embodiment is shown in Figure 1:

FIG. 1



Id., Fig. 1.

38. Ikeda's system forms a main microphone signal V_m which "has directional characteristics on the front surface side, and therefore has directional characteristics for picking up a target sound and noise from the front surface side."

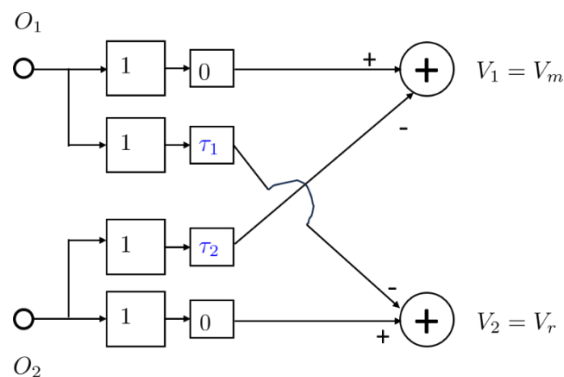
Id., ¶ (0021). V_m is formed by subtracting the output of M2 delayed by τ_2 from the output of M1 at the inversion adding circuit 3. *Id.*, ¶ (0018).

39. Similarly, reference microphone V_r which "has directional characteristics on the back surface side, and therefore has directional characteristics for picking up noise from the back surface side without picking up the target sound and noise from the front surface side." *Id.*, ¶ (0023). V_r is formed by subtracting the

output of M1 delayed by τ_1 from the output of M2 at the inversion adding circuit 4.
Id., ¶ (0018).

40. Ikeda teaches that noise cancellation is performed “[b]y determining the difference between the output signal component of the main microphone and the output signal component of the reference microphone.” *Id.*, ¶ (0012). The V_m signal is digitized in the first A/D conversion means (5), and the V_r signal is digitized in the second A/D conversion means (6). *Id.*, ¶ (0016). A digital signal processor is provided which includes “an adaptive filter means (7) for receiving the output of the second A/D conversion means (6); and a digital adding means (8) for calculating the difference between the output of the adaptive filter means and the output of the first A/D conversion means.” *Id.*, ¶ (0017).

41. The configuration of the Ikeda circuit based on Figure 1 of Ikeda and the supporting disclosure can be shown as follows:



Circuit of Ikeda, Figure 1

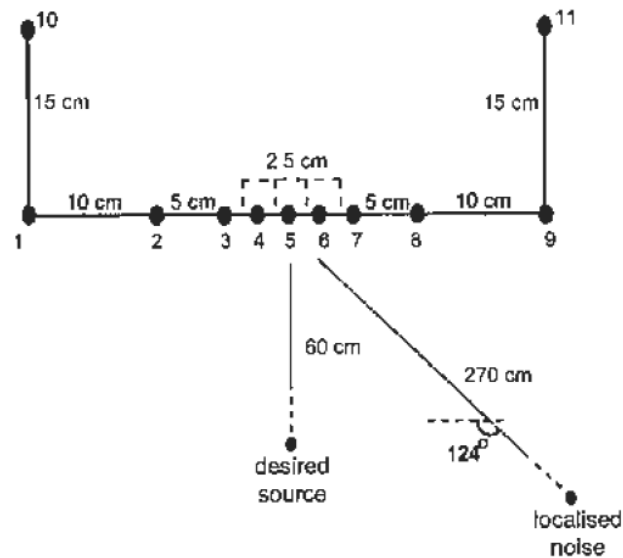
In this diagram, the signals from O_1 and O_2 are not subject to a gain. As a result, the gains shown are assigned a value of 1, signifying no gain. The signals pass through a delay τ_1 , τ_2 , or 0 (i.e., no delay). The resulting signals are ultimately summed to form V_m and V_r .

42. Ikeda does not disclose any application in near-field signal processing. I agree with Dr. Viperman that Ikeda assumes that the sources of the target signal and noise are in the far-field of the microphones. Ex. 1003, ¶ 61. Among the far-field applications disclosed by Ikeda are “a voice input microphone of a hands-free telephone unit or a car navigation system.” Ex. 1006, ¶ (0036).

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)

43. I have reviewed *Near-Field Adaptive Beamformer for Robust Speech Recognition* (“McCowan”) (Ex. 1007). McCowan was purportedly published in 2002.

44. 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. 1007 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 124° and a distance of 270 cm from the array.
Id. at 96.

45. 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:

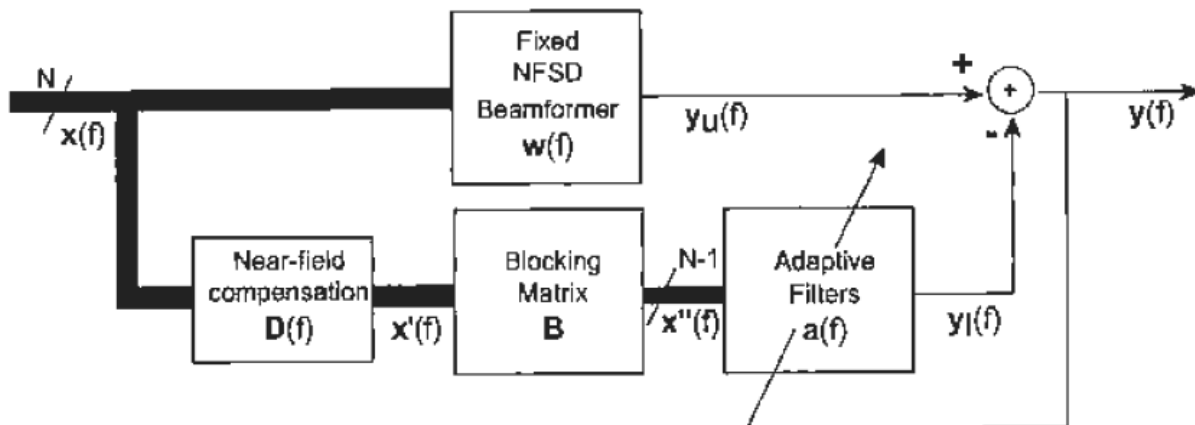


FIG. 3. Near-field adaptive beamformer.

Ex. 1007 at 91, Fig. 3.

46. 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.

47. 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.

48. 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:

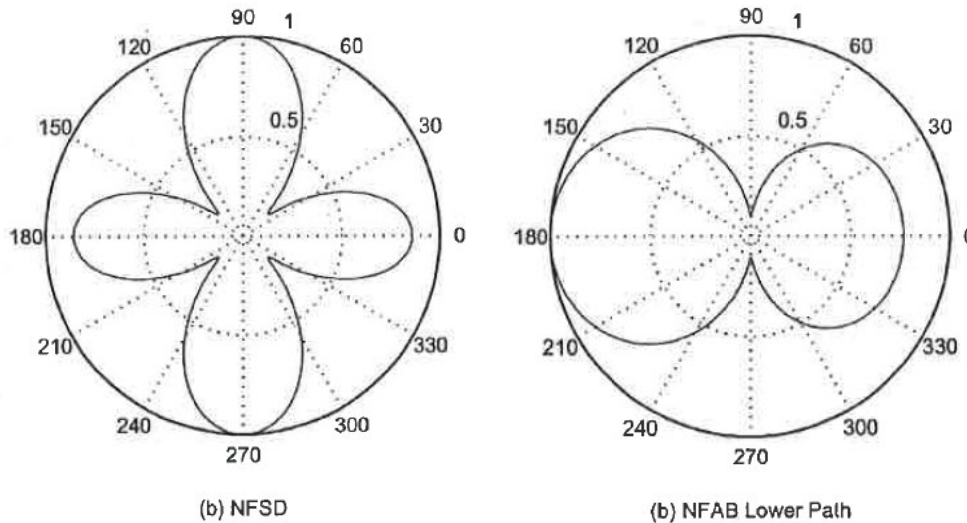


FIG. 6. Upper path directivity pattern at 300 Hz. FIG. 7. Lower path directivity pattern at 300 Hz.

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

49. 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.

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

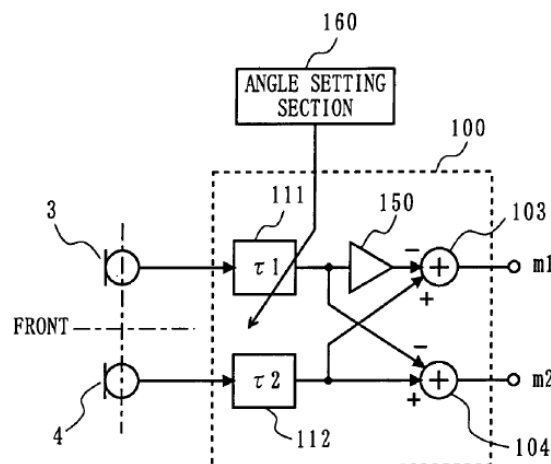
50. I have reviewed U.S. Patent Application Publication No. 2004/0185804 (“Kanamori”) (Ex. 1008). Kanamori was filed on November 18, 2003, and published

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. 1008, ¶ (0002).

51. 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.*

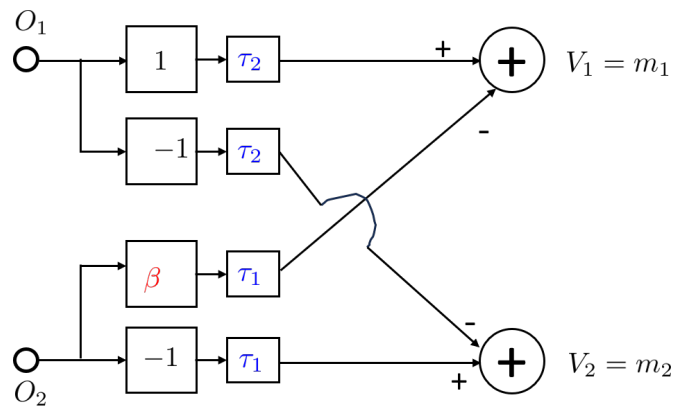
52. Kanamori discusses at least 11 separate embodiments of its system, variously comprising up to six microphones. *Id.*, ¶ (0192). Embodiment 8, on which the Petition relies, is shown below:

FIG. 17A



Ex. 1008, Fig. 17A.

53. 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. 1008, 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 m_1 and m_2 .

54. 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. 1008, ¶ (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. 1008, ¶ (0175). According to Kanamori, “angle setting section 160 can change a signal delay amount τ_1 of the first signal delaying section 111 in a range of $0 \leq \tau_1 \leq 2d/c$ (where d is a distance between the microphone units and c is the speed of sound).” Ex. 1008, ¶ (0175). When the second signal delaying section 112 includes a delay amount τ_2 set as $\tau_2 = d/c$, the full adjustment range of ± 90 degrees is achieved. *Id.*

55. Kanamori’s system in Embodiment 8, which uses the same microphone arrangement as Embodiment 6, Ex. 1008, ¶¶ (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. 1008, ¶¶ (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. 1008, Fig. 9.

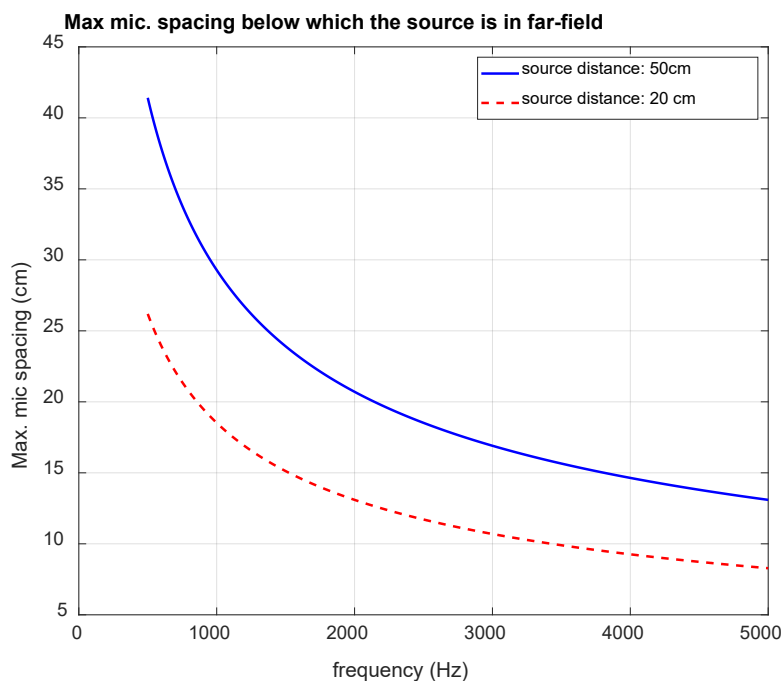
56. McCowan states that the near-field cut off for the distance (r) between a source and a microphone (antenna) array is

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

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 $\lambda = c/f$, 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. 1007 at 89.

57. 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.

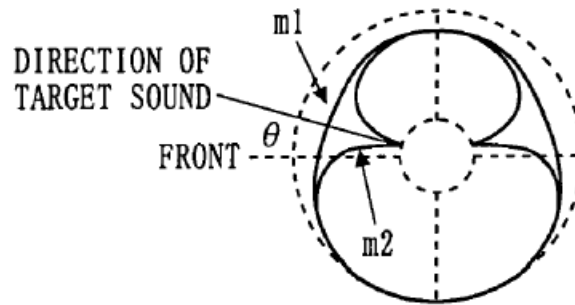


As such, Kanamori assumes that speech and the accompanying noise are in the far field.

58. 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. 1008, ¶ (0019). Consistent with that approach, Kanamori’s noise reference signal (“m2”) in Embodiment 8 includes two nulls:

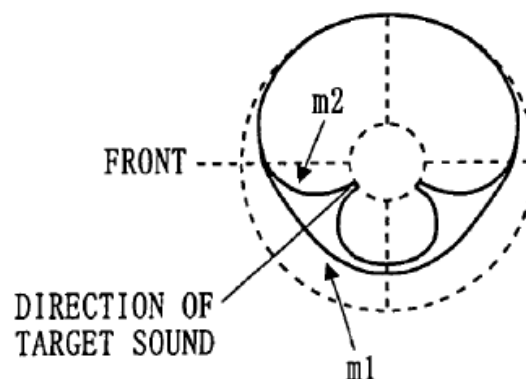
F I G. 1 7 B



Ex. 1008, Fig. 17B

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

F I G. 1 7 C



Ex. 1008, Fig. 17C.

60. Kanamori teaches that in “the direction of the target sound, there is a large difference in sensitivity between the signals m1 and m2.” Ex. 1008, ¶ (0168).

In other directions, “it is preferable that there is no difference insensitivity [sic] therebetween.” *Id.*

D. Yang (U.S. Patent Application Publication No. 2002/0193130)

61. I have reviewed U.S. Patent Application Publication No. 2002/0193130 (“Yang”) (Ex. 1009). Yang was filed on February 12, 2002, and published on December 19, 2002. Yang is directed to “[t]echniques to suppress noise from a signal comprised of speech plus noise.” Ex. 1009 at Abstract.

62. Yang’s noise suppression unit 230b uses an adaptive filter 450 to filter a noise signal before it is subtracted from a speech plus noise signal. *Id.*, ¶ (0059); Fig. 4. The noise suppression unit 230b also includes a voice activity detector 440 that “detects the presence of speech activity . . . and provides a control signal that enables the adaption of the coefficients of the base filter when no speech activity is detected.” *Id.*, ¶ (0061).

VIII. CLAIM CONSTRUCTION

63. 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 ’080 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 ’080 Patent’s disclosure and prosecution history.

IX. PETITIONER'S COMBINATIONS DO NOT RENDER ANY CLAIM OBVIOUS

A. GROUND 1: The Combination of Ikeda in View of McCowan and Kanamori Does Not Render Obvious Claims 1-3, 5-9, 11-14, and 16-20

64. In my opinion, the combination of Ikeda, McCowan, and Kanamori does not render claims 1-3, 5-9, 11-14, and 16-20 of the '080 Patent obvious.

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

65. In my opinion, the Petition contains no evidence that a POSITA would have been motivated to combine Ikeda, McCowan, and Kanamori as set forth in the Petition. The Petition's obviousness analysis is based on a hindsight reverse engineering of the noise suppression device of the '080 Patent.

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

66. The Petition admits that Ikeda is a far-field system, i.e., the system assumes that the target sound and noise are in the far field. Ex. 1003, ¶ 61 (citing Ex. 1006, ¶¶ (0002)-(0004), (0020)-(0024), (0036), Figs. 2-7). In my opinion, the Petition cites no evidence that Ikeda teaches or suggests that the disclosed noise suppression system should be adapted for use in a device to reduce noise in a target signal in the near field.

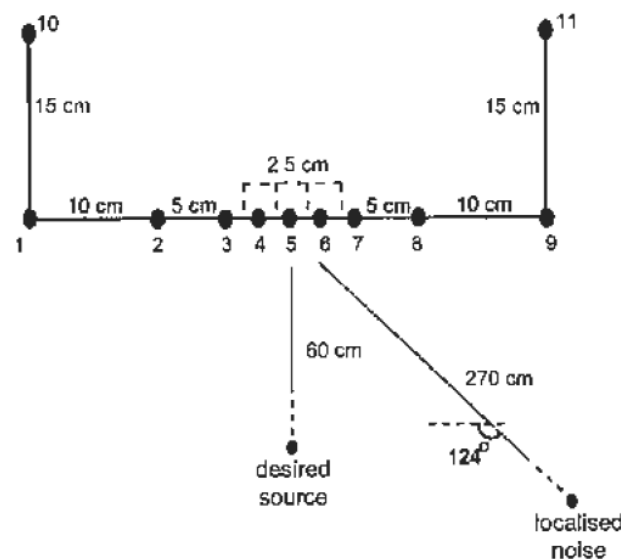
67. The Petition’s motivation to modify Ikeda for implementation in a near-field system generally, and a headset specifically, is that a POSITA would recognize that a headset is a near-field alternative to the disclosed far-field application of a “voice input microphone of a hands-free telephone unit.” Ex. 1003, ¶ 61. I disagree that even if a POSITA would recognize that a headset is a “voice input microphone of a hands-free telephone unit,” a POSITA would have been motivated to modify Ikeda for near-field application based merely on that recognition. Although the Petition identifies references supposedly disclosing that a headset is a near-field version of a hands-free telephone unit, Pet. at 20-21, the Petition does not argue that any of these references would have motivated a POSITA to modify Ikeda to make a headset. The Petition does not explain why a POSITA would have chosen a headset among the myriad unidentified applications that Ikeda’s system could have supported, much less those which would have required a POSITA to modify Ikeda.

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

68. In my opinion, even if a POSITA would have been motivated to modify Ikeda to make a headset, the Petition presents no evidence or explanation why a POSITA would have been motivated to combine Ikeda with McCowan to achieve that modification. The Petition does not explain why a POSITA would have modified Ikeda to add a gain to the Ikeda circuit in view of the teachings of

McCowan. Even if one assumed that adding a gain to Ikeda's Vr circuit would have improved the circuit, Ex. 1003, ¶ 62, the Petition nevertheless fails to articulate why a POSITA would have looked to McCowan for that improvement or would have combined McCowan with Ikeda in the specific way described by the Petition.

69. 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. 1007 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.

70. 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. 1007 at 87, 104. The Petition does not purport to modify

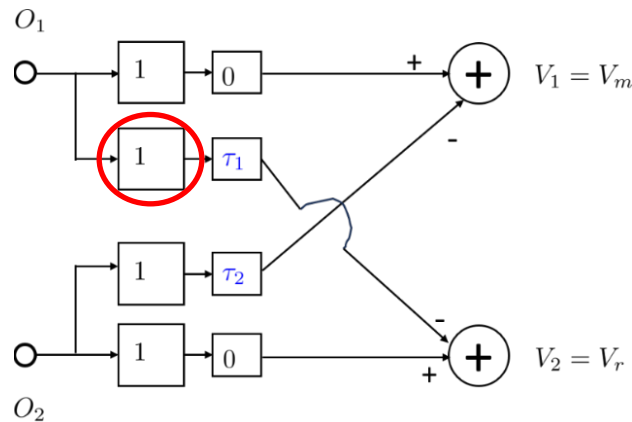
Ikeda to make a headset with improved noise suppression at low frequencies rather than across the entire frequency band for speech.

71. The Petition presents no evidence why a POSITA would have looked to McCowan to provide a component to modify Ikeda to make a headset. The Petition also does not identify the motivation to create a headset as the reason it selected McCowan to modify Ikeda. In my review of the Petition, I could find no reference to a headset, or the motivation to create a headset, mentioned in connection with the combination with McCowan or Kanamori, or for any other purpose.

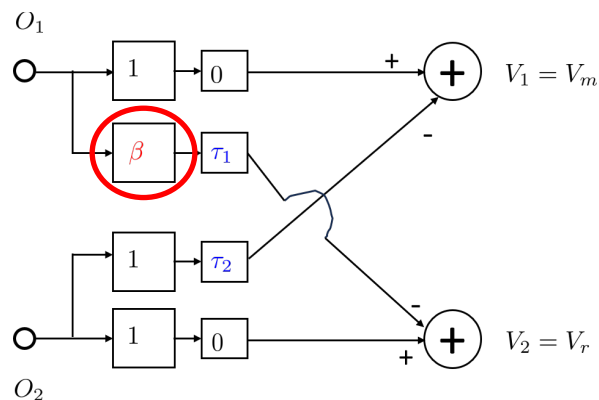
72. In my opinion, the Petition's selection of a single gain from the McCowan circuit to modify Ikeda while ignoring other aspects of McCowan's apparatus is evidence of hindsight. The Petition argues that adding a gain to Ikeda's V_r circuit would have been an improvement, but the Petition does not state why a POSITA would not have incorporated into Ikeda other aspects of McCowan, such as the fixed near-field superdirective beamformer that processes the upper path of McCowan's apparatus. Ex. 1007, Fig. 3. Since McCowan is a near-field reference and Ikeda is not, a POSITA also would have considered modifying Ikeda's V_m circuit according to McCowan's disclosure. The Petition is silent regarding why a POSITA was motivated to modify Ikeda's V_r circuit in view of McCowan but not Ikeda's V_m circuit.

73. In my opinion, a comparison of the Ikeda-McCowan combination's

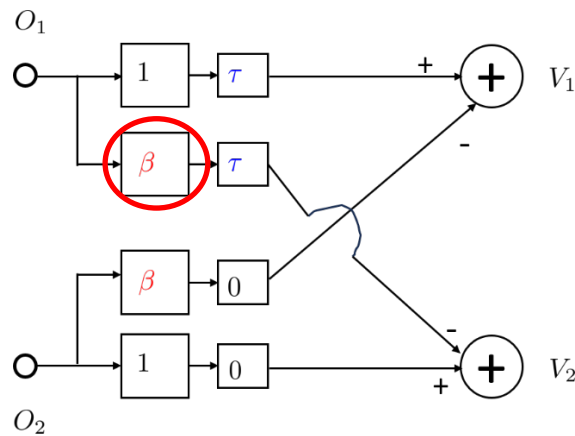
circuit and the '080 Patent circuit reveals that a gain was incorporated is required for Petitioner to replicate the '080 Patent's V_2 circuit. Following is the circuit disclosed in Ikeda showing no gain in the second line:



Following is the circuit resulting from the Ikeda-McCowan combination in the Petition with a delay added to the second row of the circuit:



Finally, following is the circuit disclosed in the '080 Patent:



The second and fourth lines of these circuits form the inputs to the adders that create the V_r (V_2) signals in the respective circuits.

74. The only difference between the second and fourth lines of the circuits of Ikeda and '080 Patent is the lack of a gain in the second line of Ikeda. Petitioner addressed this difference by incorporating the gain of McCowan into the second line of Ikeda. Although the Petition characterized this change as an improvement to Ikeda, Pet. at 21, the Petition provided no explanation why it did not incorporate the remaining teachings of McCowan into the first and third lines of the Ikeda-McCowan combination to improve the V_m (V_1) circuit, such as by incorporating McCowan's fixed near-field superdirective beamformer to improve the quality of the V_m (V_1) signal in Ikeda. In my opinion, this is evidence of the exercise of hindsight.

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

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

76. The Kanamori embodiment that Petitioner combines with the Ikeda-McCowan circuit 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. 1008, ¶¶ (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. 1008, Fig. 9. In this configuration, the talker is in the far field. *See supra.*, ¶¶ 55-57.

77. The Petition does not explain why a POSITA would combine Ikeda as modified by McCowan with a far-field device such as Kanamori when the motivation to make the combination was to create a near-field device. McCowan was chosen for combination with Ikeda because it discloses a near-field application. Pet. at 22, 23. Petitioner looks to Kanamori to modify the main microphone of the Ikeda-McCowan combination to improve denoising of speech in the near field even

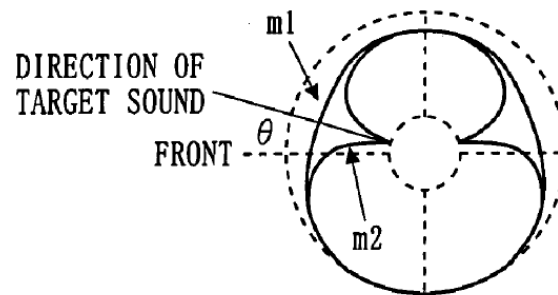
though Kanamori's main microphone is used for denoising speech in the far field. The Petition does not address the inherent inconsistency in these positions.

iv. The Petition's Argument that Kanamori Teaches that the Directivity Patterns of Two Virtual Microphones Be as Similar as Possible Is Incorrect

78. In my opinion, Kanamori does not teach that a "the directivity patterns of main signal m1 and noise reference signal m2 are preferably as similar as possible in directions from which noise is coming." Pet. at 27.

79. In Embodiment 7, Kanamori teaches that there should be a large difference in sensitivity between the signals m1 and m2 in the direction of the target sound. Ex. 1008, ¶ (0168). But, it is preferable that there be "no difference in sensitivity" in "directions other than the direction of the target sound." *Id.* The reason is that canceling noise by subtracting m2 from m1 without also cancelling the target signal requires that the sensitivity of m2 in the direction of the target speech be as different from m1 as possible as shown in Figure 16B:

F I G. 1 6 B



Ex. 1008, Fig. 16B.

80. I disagree with the Petition’s argument that Kanamori discloses that “the directivity patterns of main signal m1 and noise reference signal m2 are preferably as similar as possible in directions from which noise is coming.” Kanamori is a far-field system where both the target signal and noise are in the far field. Noise can originate in any direction, including in the direction of the target sound. Kanamori teaches that a difference in the sensitivity in the direction of the target sound is necessary to avoid cancelling out the target sound. This is true even if there is noise originating in the direction of the target sound. Thus, Kanamori’s teaching is agnostic as to the direction from which noise is coming. Rather, Kanamori is focused on maintaining a difference in sensitivity in the direction of the target sound, even if noise originates in the same direction, and as close to identical sensitivity in other directions.

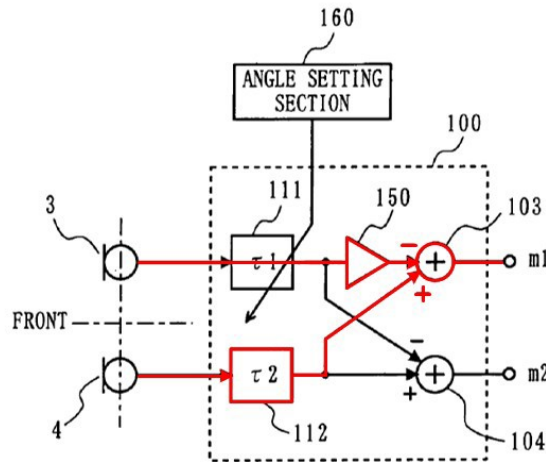
81. The Petition's mischaracterization of Kanamori is significant because it forms the motivation for the Petition's reverse engineering of the Vr (V2) circuit in Ikeda-McCowan to incorporate circuitry from Kanamori to arrive at the '080 Patent circuit. The Petition relies on Kanamori's alleged teaching that it accurately estimates noise "irrespective of the direct from which noise is coming" when, in fact, as shown in Figure 16B, the noise reference signal does not accurately estimate noise in the direction of the target sound and in one other direction as shown in Figure 16B of Kanamori. As such, the Petition's motivation to combine Kanamori with Ikeda-McCowan to improve the main signal Vm using Kanamori's m2 is not valid.

v. The Petition's Alterations to Kanamori to Arrive at the Combination Discards the Inventive Concept of the Kanamori Embodiment without Explanation

82. In my opinion, the Petition discards the inventive concept of the Kanamori embodiment on which it relies without explanation so that it may reverse engineer the '080 Patent's circuit.

83. The Petition states that the following configuration for generating main signal m1 should be used in combination with Ikeda-McCowan:

FIG. 17A



Pet. at 29. In this revised configuration (the revisions are shown in red), the Petition advocated for setting τ_1 in the first signal delaying section 111 to zero, which effectively removes that component from the system as shown by the red line running through item 111. Pet. at 29. The Petition offers no reason to modify the Kanamori circuit, stating that the modification “would be readily applicable to generating the main signal V_m in Ikeda-McCowan.” Pet. at 29.

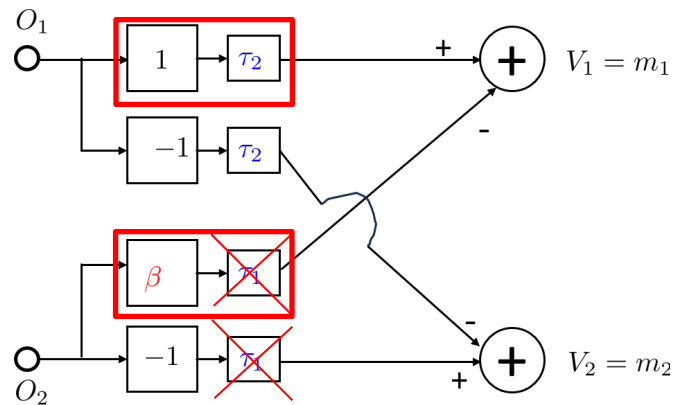
84. In my opinion, a POSITA would not have been motivated to make Petitioner’s modification because it removes the inventive aspect of Figure 17A. Kanamori discloses that Embodiment 8 shown in Figure 17A improves on Embodiment 7 by adding an angle setting section 160 and a second signal delaying section 112. Ex. 1008, ¶ (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. 1008, ¶ (0175). According to Kanamori, “angle setting section 160 can change a

signal delay amount τ_1 of the first signal delaying section 111 in a range of $0 \leq \tau_1 \leq 2d/c$ (where d is a distance between the microphone units and c is the speed of sound).” Ex. 1008, ¶ (0175). When the second signal delaying section 112 includes a delay amount τ_2 set as $\tau_2 = d/c$, the full adjustment range of ± 90 degrees is achieved. *Id.*

85. In my opinion, removing the first signal delaying section 111 and, therefore, the angle setting section 160 eviscerates the purpose of Embodiment 8, which was to provide an adjustable τ_1 and include a second delay τ_2 .

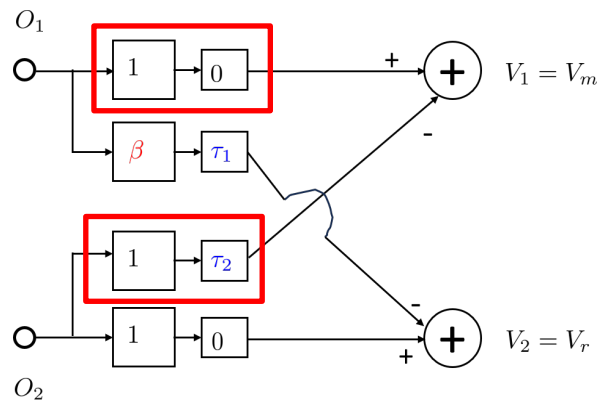
86. In my opinion, Petitioner was motivated to eliminate τ_1 from Kanamori so that the combination with Ikeda-McCowan would arrive at the '080 Patent circuit.

87. The Petition proposes to modify the main microphone of the Ikeda-McCowan combination based on the revised Kanamori circuit of Figure 17A shown below. The main microphone m_1 in Kanamori is generated by the first and third lines of the circuit shown in boxes below. The circuit is annotated with an “x” through the delay, τ_1 , that Petitioner seeks to eliminate:



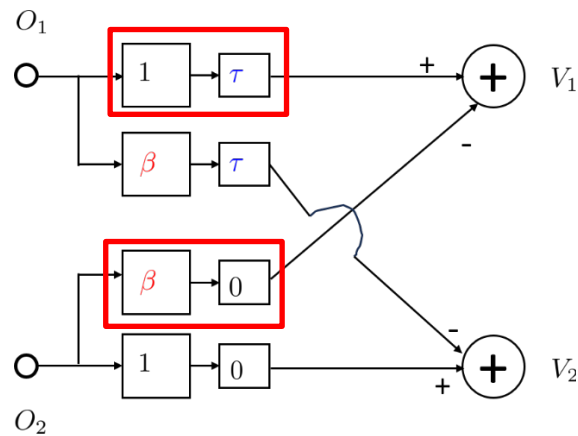
Kanamori Circuit of Fig 17A (annotated)

88. Petitioner uses revised Kanamori Figure 17A to modify the main microphone V_m of Ikeda-McCowan which is generated by the first and third lines of the circuit, shown below:



Ikeda-McCowan Circuit (annotated)

89. Following is the '080 Patent circuit that Petitioner is attempting to replicate:



'080 Patent Circuit (annotated)

Note that the '080 Patent circuit does not include a delay in the third line of the circuit.

90. These diagrams show that Petitioner was motivated to eliminate τ_1 from the Kanamori circuit because the '080 Patent circuit does not use a delay in the third line of the circuit. Additionally, the Ikeda-McCowan circuit teaches use of a delay in the third line of the circuit, but Petitioner eliminates it from Ikeda-McCowan without explanation. Petitioner's simulation of the combined Ikeda-McCowan-Kanamori circuit would not yield the linear responses shown in the Petition had Petitioner not revised the Kanamori specification for the purpose of replicating V_1 in the '080 Patent circuit. Pet. at 44-46, 49-54.

X. CONCLUSION

91. 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 21st day of September, 2023

Respectfully submitted,

A handwritten signature in black ink, appearing to read "Mark S. Saylor", is written over a horizontal line.