

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

SONY ELECTRONICS INC.,
Petitioner,

v.

JAWBONE INNOVATIONS, LLC,
Patent Owner.

IPR2023-01065
Patent 10,779,080 B2

Before GEORGIANNA W. BRADEN, SHEILA F. McSHANE, and
JASON M. REPKO, *Administrative Patent Judges*.

BRADEN, *Administrative Patent Judge*.

DECISION
Granting Institution of *Inter Partes* Review
35 U.S.C. § 314

I. INTRODUCTION

Sony Electronics, Inc. (“Petitioner”) filed a Petition requesting an *inter partes* review of claims 1–20 (the “challenged claims”) of U.S. Patent No. 10,779,080 B2 (Ex. 1001, “the ’080 patent”). Paper 1 (“Pet.”). Jawbone Innovations, LLC (“Patent Owner”) filed a Preliminary Response. Paper 8 (“Prelim. Resp.”). With our permission, Petitioner filed a Reply, to address arguments regarding “near-field” and “far-field” issues raised in previous proceedings. Paper 9 (“Reply”). Patent Owner filed a Sur-reply. Paper 10 (“Sur-reply”).

We have authority to determine whether to institute an *inter partes* review under 35 U.S.C. § 314 and 37 C.F.R. § 42.4. An *inter partes* review may not be instituted unless it is determined that “the information presented in the petition filed under section 311 and any response filed under section 313 shows that there is a reasonable likelihood that the petitioner would prevail with respect to at least 1 of the claims challenged in the petition.” 35 U.S.C. § 314 (2018); *see also* 37 C.F.R. § 42.4(a) (2022) (“The Board institutes the trial on behalf of the Director.”). The reasonable likelihood standard is “a higher standard than mere notice pleading,” but “lower than the ‘preponderance’ standard to prevail in a final written decision.” *Hulu, LLC v. Sound View Innovations, LLC*, IPR2018-01039, Paper 29 at 13 (PTAB Dec. 20, 2019) (precedential).

For the reasons provided below and based on the record before us, we determine Petitioner has demonstrated a reasonable likelihood that it would prevail in showing the unpatentability of at least one of the challenged claims. Accordingly, we institute an *inter partes* review of the ’080 patent and all of the asserted challenges to patentability.

II. BACKGROUND

A. Real Parties in Interest

Petitioner lists Sony Electronics, Inc., Sony Group Corp., Sony Corporation, and Cony Corporation of America as a real parties-in-interest. Pet. 90. Patent Owner states that it is a “real party-in-interest.” Paper 6 (Patent Owner’s Mandatory Notice), 2.

B. Related Proceedings

The parties both identify the following multiple district court cases involving the ’080 patent as well as related IPR proceedings: IPR2022-01059; IPR2022-01320; and IPR2023-00284. Pet. 86; Paper 6, 2–3. Each of the related IPR proceedings have been terminated due to settlement. *See* IPR2022-01059, Paper 27; IPR2022-01320, Paper 18; IPR2023-00284, Paper 16.

C. Motion for Joinder and Consolidation

Petitioner moves for joinder to and consolidation with related IPR proceeding IPR2023-00284 under 35 U.S.C. § 315 and 36 C.F.R. § 42.122(b). Paper 3 (Motion for Joinder). IPR proceeding IPR2023-00284 has been terminated due to settlement. *See* IPR2023-00284, Paper 16. Thus, there is no proceeding with which to join the present case. Accordingly, Petitioner’s motion for joinder and consolidation is denied.

D. The ’080 Patent (Ex. 1001)

The ’080 patent is titled “Dual Omnidirectional Microphone Array (DOMA),” and issued on September 15, 2020. Ex. 1001, codes (45), (54). It is a continuation of a U.S. Patent Application and relies on several provisional applications, the earliest of which was filed on June 13, 2007. *Id.* at codes (60), (63).

1. *Written Description*

The '080 patent relates to the use of a “dual omnidirectional microphone array noise suppression” system to create microphone arrays with enhanced noise suppression.” *Id.* at code (57). The '080 patent explains that the

array of an embodiment is used to form two distinct virtual directional microphones which are configured to have very similar noise responses and very dissimilar speech responses. The only null formed is one used to remove the speech of the user from V2. The two virtual microphones may be paired with an adaptive filter algorithm and VAD algorithm to significantly reduce the noise without distorting the speech, significantly improving the SNR of the desired speech over conventional noise suppression systems.”

Id. According to 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. The similarity of the '080 patent’s virtual microphones’ noise responses, however, 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; *see id.* at 9:22–38. The '080 patent discloses that its two virtual microphones can be “paired with an adaptive filter algorithm” to reduce noise. *Id.* at 3:57–62.

In one embodiment, two virtual directional microphones V_1 and V_2 are formed based on two physical omnidirectional microphones O_1 and O_2 according to the following equations expressed in the z (digital frequency) domain:

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

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

where

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

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

$$d_1 = \sqrt{d_s^2 - 2d_{s1}d_0\cos(\theta) + d_0^2}$$

$$d_2 = \sqrt{d_s^2 + 2d_{s1}d_0\cos(\theta) + d_0^2}$$

Ex. 1001, 4:33–67, 9:46–53, 11:1–2. In the equations d_1 and d_2 “are the distance from O_1 and O_2 to the speech source,” respectively; c is the speed of sound; and f_s is a sampling frequency. *Id.* at 9:54–67. The equations for V_1 and V_2 (above) are represented visually in Figure 2, reproduced below.

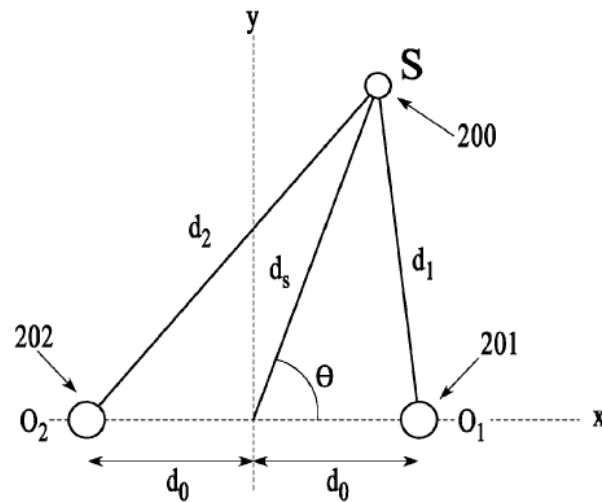


FIG.2

Figure 2 is a diagram illustrating a generalized two-microphone array (DOMA) including array 201/202 and speech source (S) with

microphones separated by a distance approximately equal to $2d_o$, and the speech source located a distance d_s away from the midpoint of the array at an angle θ . According to the '080 patent, the system is axially symmetric so only d and θ need be specified. *Id.* at 2:28–34, 7:17–38, Fig. 2.

The result of one embodiment of the '080 patent is shown in Figure 9, reproduced below.

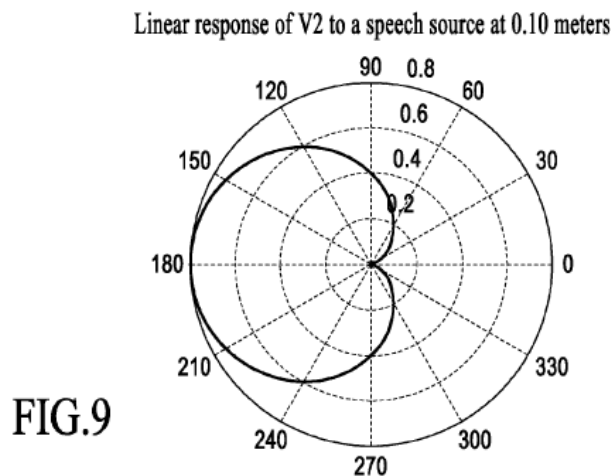
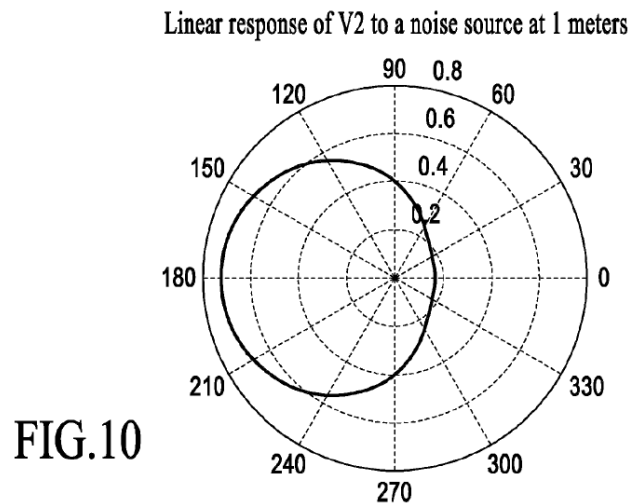


Figure 9 is a plot of linear response of virtual microphone V_2 with $\beta=0.8$ to a 1 kHz speech source at a distance of 0.1 m. *Id.* at 10:18–20. According to the '080 patent, the null in the linear response of virtual microphone V_2 to speech is located at 0 degrees, where the speech is typically expected to be located. *Id.* at 10:20–22. Figure 10, reproduced below, is another plot of a linear response of virtual microphone V_2 with $\beta=0.8$ to a 1 kHz noise source, but at a distance of 1.0 m. *Id.* at 10:22–24.



Regarding Figure 10, above, the '080 patent states that “[t]he linear response of V_2 to noise is devoid of or includes no null, meaning all noise sources are detected.” *Id.* at 10:25–27.

The '080 patent discloses that the formulation for $V_2(z)$ has a null at the speech location, and therefore, will exhibit minimal response to the speech. *Id.* at 10:28–30. The '080 patent then explains that the speech null at zero degrees is not present for noise in the far field for the same microphone, as shown in Figure 10, above, with a noise source distance of approximately 1 meter. *Id.* at 10:32–35. The '080 patent states this insures that noise in front of the user will be detected so that it can be removed. *Id.* at 10:35–36. According to the '080 patent, this differs from conventional systems that can have difficulty removing noise in the direction of the mouth of the user. *Id.* at 10:36–38.

The '080 patent further discloses different linear responses for the virtual microphone V_1 and V_2 when exposed to a near-field speech source. *Id.* at 10:18–22, 11:20–24, 11:62–65, 16:4–6. This difference is seen when comparing Figures 9 and 11, both reproduced below.

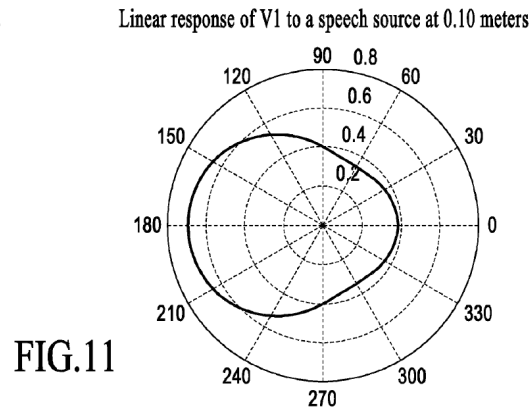
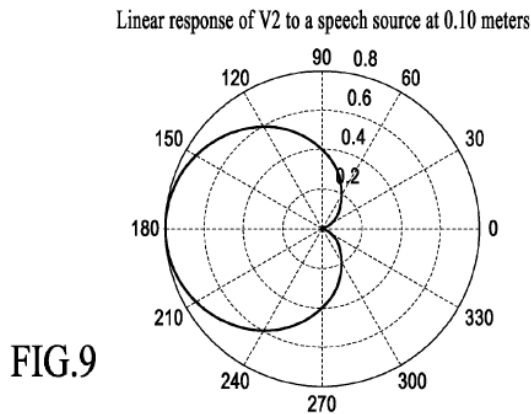


Figure 11 is a plot of linear response of virtual microphone V_1 to a 1 kHz speech source at a distance of 0.1 m. Ex. 1001, 2:60-62. The '080 patent states that “[t]here is no null and the response for speech is greater than that shown in Figure 9.” *Id.* at 2:62–63.

In contrast, the '080 patent describes similar linear responses for virtual microphones V_1 and V_2 , respectively, to a far-field noise source. *Id.* at 10:22–27, 11:25–29, 11:62–65, 16:4–6. This similarity is seen when comparing Figures 10 and 12, both reproduced below.

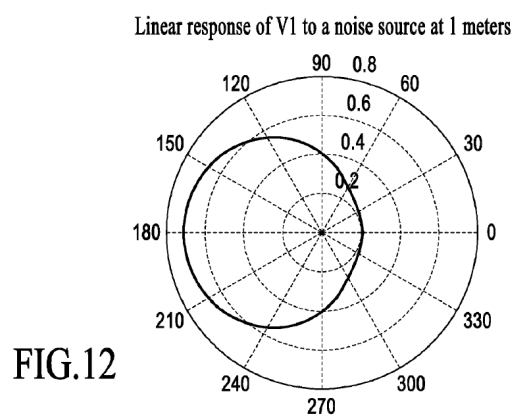
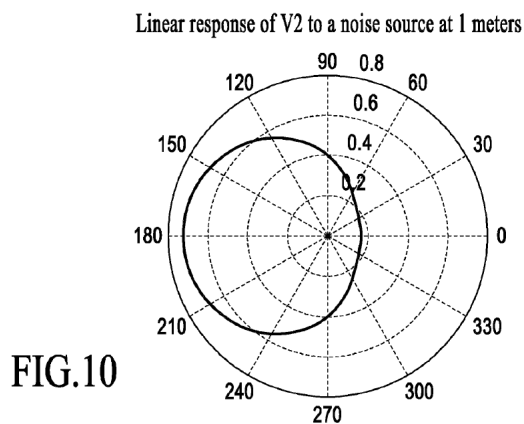


Figure 12, above, is a plot of linear response of virtual microphone V_1 to a 1 kHz noise source at a distance of 1.0 m, under an embodiment. *Id.* at 2:64–66. The '080 patent states that “[t]here is no null and the response is very similar to V_2 shown in Figure 10.” *Id.* at 2:66–67.

The '080 patent contemplates that the linear response of the virtual microphone V_1 is “flat for all frequencies within approximately ± 30 degrees of the axis of the array,” as shown in Figure 13 (reproduced below).
Ex. 1001, 11:30–35, 11:43–61;

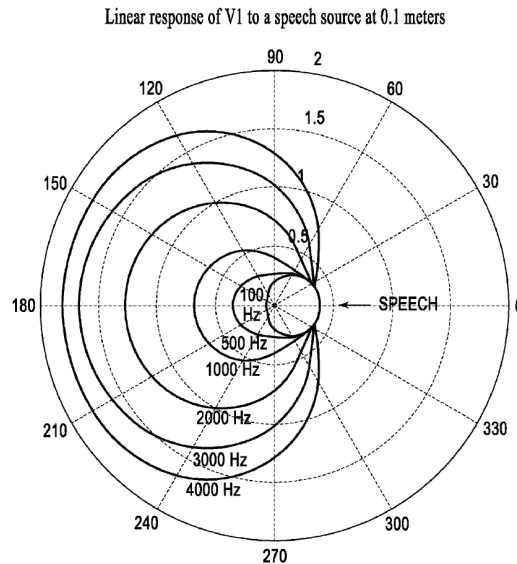


FIG.13

“Figure 13 is a plot of linear response of virtual microphone V_1 to a speech source at a distance of 0.1 m for frequencies of 100, 500, 1000, 2000, 3000, and 4000 Hz.” *Id.* at 3:1–3.

The virtual microphone signals are input to an adaptive noise removal system, which can use “any adaptive filter or noise removal algorithm.” *Id.* at 3:57–62, 6:67–7:3, 8:50–9:3, 9:22–38; *see id.* at 4:52–7:16. In an embodiment, the adaptive noise removal system combines the signals from virtual microphones V_1 and V_2 “by filtering and summing in the time domain.” *Id.* at 7:4–16, 8:50–9:3, 9:22–38. “The adaptive filter generally uses the signal received from a first microphone . . . to remove noise from the speech received from at least one other microphone . . . , which relies on

a slowly varying linear transfer function between the two microphones for sources of noise.” *Id.* at 7:7–12. The system then generates an output signal “in which the noise content is attenuated with respect to the speech content.” *Id.* at 7:12–16.

2. *Illustrative Claims*

As noted previously, Petitioner challenges claims 1–20 of the ’080 patent, of which claims 1, 7, and 14 are independent. Pet. 2; Ex. 1001, 33:39–36:25. Claim 7 is illustrative of the challenged subject matter and is reproduced below.

7. A method comprising:

- a first virtual microphone formed from a first combination of a first microphone signal and a second microphone signal, wherein the first microphone signal is generated by a first physical microphone and the second microphone signal is generated by a second physical microphone;
- a second virtual microphone formed from a second combination of the first microphone signal and the second microphone signal, wherein the second combination is different from the first combination, wherein the first virtual microphone has a first linear response to speech and first linear response to noise, the first linear response to speech being substantially similar across a plurality of frequencies for a speech source located within a predetermined angle relative to an axis of the microphone array and devoid of a null, wherein the second virtual microphone has a second linear response to speech that has a single null oriented in a direction toward a source of the speech and a second linear response to noise, wherein the second linear response to noise is substantially similar to the first linear response to noise, one or both of the first linear response to noise and the second linear response to noise being non-zero in a direction toward a source of noise, and the second linear response to speech is substantially dissimilar to the first linear response to speech, wherein the speech is human speech; and
- an adaptive noise removal application coupled to the first and second virtual microphones and generating denoised output signals by

forming a plurality of combinations of signals output from the first virtual microphone and the second virtual microphone, by filtering and summing the plurality of combinations of signals in the time domain, and by a varying linear transfer function between the plurality of combinations of signals, wherein the denoised output signals include less acoustic noise than acoustic signals received at the first and second physical microphones.

Ex. 1001, 34:16–54.

E. Asserted Challenges to Patentability and Evidence of Record

Petitioner challenges the patentability of claims 1–20 of the '080

patent based on the following reference or combination of references:

Claims Challenged	35 U.S.C. §	Reference(s)/Basis
1–3, 5–9, 11–14, 16–20	103 ¹	Ikeda ² , McCowan ³ , Kanamori ⁴
4, 10, 15	103	Ikeda, McCowan, Kanamori, Yang ⁵

¹ The Leahy-Smith America Invents Act, Pub. L. No. 112-29, 125 Stat. 284 (2011) (“AIA”), included revisions to 35 U.S.C. § 103 that became effective as of March 16, 2013. Although the application for the '080 patent was filed after March 16, 2013, it includes a priority claim to an application filed before this date. Ex. 1001, codes (22), (60), (63). Accordingly, for purposes of institution, we apply the pre-AIA version of 35 U.S.C. § 103.

² Japanese Unexamined Patent Application Publication No. H11-18186, filed June 25, 1997, published January 22, 1999; (“Ikeda,” Ex. 1006).

³ Iain A. McCowan et al., *Near-Field Adaptive Beamformer for Robust Speech Recognition*, Digital Signal Processing, Vol. 12, Issue 1 (2002), 87–106 (“McCowan,” Ex. 1007).

⁴ US Patent Application Publication No. 2004/0185804, published September 23, 2004 (“Kanamori,” Ex. 1008).

⁵ US Patent Application Publication No. 2002/0193130, published December 19, 2002 (“Yang,” Ex. 1009).

In support of its patentability challenge, Petitioner relies on, *inter alia*, the Declaration of Jeffrey S. Vipperman, Ph.D. (“Dr. Vipperman”). Ex. 1003.⁶

Patent Owner contests the challenges to patentability and supports its position with the Declaration of Akbar M. Sayeed, Ph.D. (“Dr. Sayeed”). Ex. 2002.

III. PRELIMINARY MATTERS

A. Claim Construction

A claim “shall be construed using the same claim construction standard that would be used to construe the claim in a civil action under 35 U.S.C. [§] 282(b).” 37 C.F.R. § 42.100(b). Under that standard, the “words of a claim ‘are generally given their ordinary and customary meaning.’” *Phillips v. AWH Corp.*, 415 F.3d 1303, 1312 (Fed. Cir. 2005) (en banc).

⁶ Dr. Vipperman was retained as an independent expert by non-party Google LLC for IPR proceeding IPR2023-00284, which challenged to the ’080 patent based on the same prior art as asserted in the present Petition. We remind the parties of their obligation under Rule 42.51(b)(1)(ii), which allows for cross examination of affidavit testimony as routine discovery. *See John’s Lone Star Distrib., Inc. v. Thermolife Int’l, Inc.*, Case No. IPR2014–01201 (PTAB May 13, 2015) (Paper 31) (petitioner submitted affidavit testimony from a witness triggering cross-examination under 37 C.F.R. § 42.51(b)(1), and because petitioner did not make that witness available for cross-examination, the Board expunged the testimony.); *IBG LLC et al. v. Trading Technologies Internationale, Inc.*, Case No. CBM2015-00179 (PTAB Apr. 15, 2016) Paper 39, 3 (The Board determined that it was the petitioner’s responsibility to produce the witness for cross-examination, because that deposition was the sole evidence that a reference was prior art.).

Both parties state that express claim construction is unnecessary at this stage of the proceeding and that all terms are to given their plain and ordinary meaning. Pet. 8; Prelim. Resp. 19. We agree.

Construction is needed only for those terms “that are in controversy, and only to the extent necessary to resolve the controversy.” *See Nidec Motor Corp. v. Zhongshan Broad Ocean Motor Co.*, 868 F.3d 1013, 1017 (Fed. Cir. 2017) (quoting *Vivid Techs., Inc. v. Am. Sci. & Eng’g, Inc.*, 200 F.3d 795, 803 (Fed. Cir. 1999)). Consequently, we need not construe any terms for purposes of institution.

B. Principles of Law Regarding Obviousness

A claim is unpatentable under 35 U.S.C. § 103 if “the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains.” *KSR Int’l Co. v. Teleflex Inc.*, 550 U.S. 398, 406 (2007). The question of obviousness is resolved on the basis of underlying factual determinations, including: (1) the scope and content of the prior art; (2) any differences between the claimed subject matter and the prior art; (3) the level of skill in the art; and (4) where in evidence, objective evidence of non-obviousness.⁷ *Graham v. John Deere Co.*, 383 U.S. 1, 17–18 (1966). When evaluating a combination of teachings, we must also “determine whether there was an apparent reason to combine the known elements in the fashion claimed by the patent at issue.” *KSR*, 550 U.S. at 418 (citing *In re Kahn*, 441 F.3d 977, 988 (Fed. Cir. 2006)). Whether a combination of prior

⁷ At this stage of the proceeding, Patent Owner has not presented objective evidence of non-obviousness.

art elements would have produced a predictable result weighs in the ultimate determination of obviousness. *Id.* at 416–417.

In an *inter partes* review, the petitioner must show with particularity why each challenged claim is unpatentable. *Harmonic Inc. v. Avid Tech., Inc.*, 815 F.3d 1356, 1363 (Fed. Cir. 2016); 37 C.F.R. § 42.104(b). The burden of persuasion never shifts to the patent owner. *Dynamic Drinkware, LLC v. Nat’l Graphics, Inc.*, 800 F.3d 1375, 1378 (Fed. Cir. 2015).

We analyze the challenges presented in the Petition in accordance with the above-stated principles.

C. Level of Ordinary Skill in the Art

In determining the level of ordinary skill in the art, various factors may be considered, including the “type of problems encountered in the art; prior art solutions to those problems; rapidity with which innovations are made; sophistication of the technology; and educational level of active workers in the field.” *In re GPAC, Inc.*, 57 F.3d 1573, 1579 (Fed. Cir. 1995) (quotation marks omitted). Furthermore, the prior art itself can reflect the appropriate level of ordinary skill in the art. *Okajima v. Bourdeau*, 261 F.3d 1350, 1355 (Fed. Cir. 2001).

Here, Petitioner asserts a person of ordinary skill in the art by June 27, 2007, would have had “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.” Pet. 8 (citing Ex. 1003 ¶¶ 22–23). Dr. Vipperman supports Petitioner’s position and further testifies that

“[w]ork experience can substitute for formal education and additional formal education can substitute for work experience.” Ex. 1003 ¶ 23.

Patent Owner does not assert a different level of skill in the art at the time of the alleged invention and uses Petitioner’s proposed level of skill. Prelim. Resp. 19 (citing Ex. 2002 ¶ 26).

For the purposes of this Decision, we too adopt Petitioner’s level of ordinary skill in the art because it appears consistent with the problems addressed in the ’080 Patent and the prior art of record.

D. Overview of Asserted Prior Art of Record

1. Ikeda (Ex. 1006)

Ikeda is an unexamined Japanese Patent Application Publication No. H11-18186, published on January 22, 1999, titled “Primary Pressure Gradient Microphone.” Ex. 1006,⁸ codes (11), (43), (54). Ikeda discloses a microphone apparatus including an adaptive noise suppression with “three omnidirectional microphone units” M1, M2, and M3. Ex. 1006 ¶¶ 8, 11–13, 16–18, 27–29, 31, 36, Fig. 8.

One embodiment of Ikeda is illustrated in Figure 1, reproduced below.

⁸ In this Decision we will refer to Ex. 1006, which is a certified translation of the original Japanese publication found at Ex. 1005.

FIG. 1

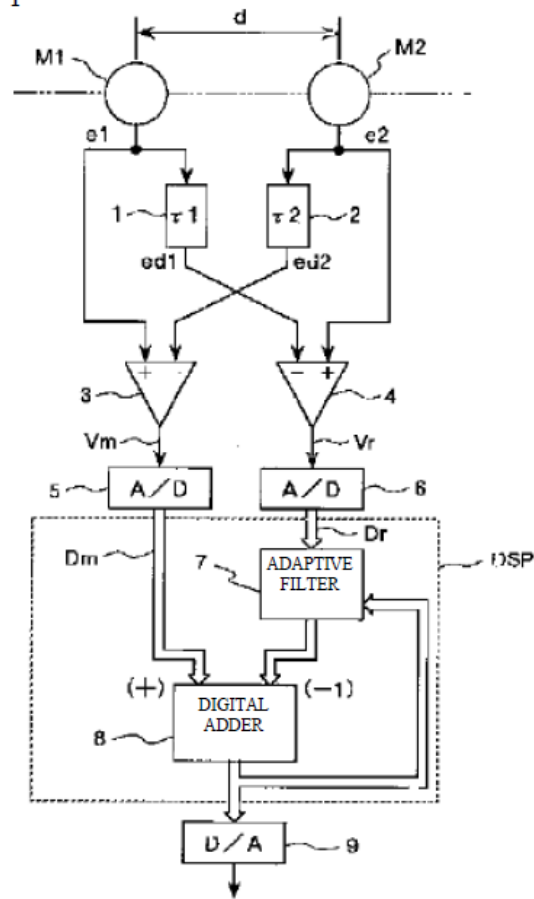


Figure 1 of Ikeda, above, is a circuit diagram of a primary pressure gradient microphone using two omnidirectional microphone units. *Id.* ¶ 11.

According to Figure 1, the output “of the microphone unit M2 is delayed . . . by a delay circuit 2,” and an inversion adding circuit 3 outputs a main signal Vm by “calculating the difference between the output . . . of the microphone unit M1 and the output . . . of the delay circuit 2.” *Id.* ¶¶ 18, 20, 31. Ikeda discloses that main signal Vm is generated by subtracting the output signal of microphone M2, as delayed by delay circuit 2, from the output signal of microphone M1. *Id.* ¶¶ 18, 21. Ikeda further discloses a signal system constructed using microphones M1 and M2, delay circuit 2, and inversion adding circuit 3 for outputting main signal Vm functions as a main

microphone. *Id.* ¶¶ 12, 20–22, Fig. 2. According to Ikeda, the main microphone has directional characteristics. *Id.* ¶¶ 2–4, 21, 22, Fig. 3.

Another embodiment of Ikeda illustrated in Figure 8, reproduced below.

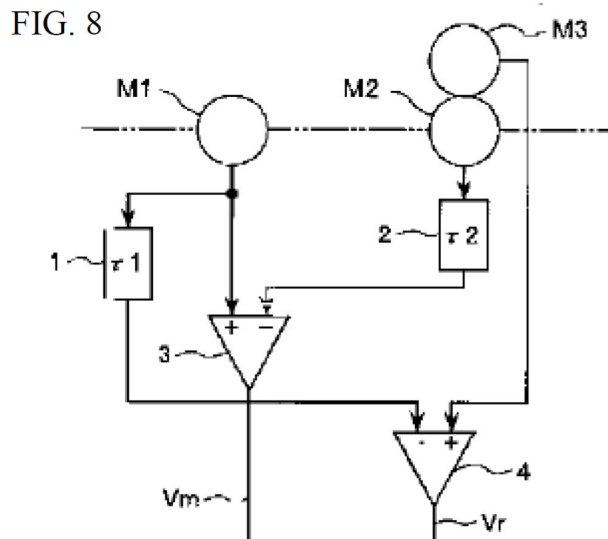


Figure 8 of Ikeda, above, is a circuit diagram of a primary pressure gradient microphone using three omnidirectional microphone units. *Id.* ¶42.

According to Figure 8, the output “of the microphone unit M1 is delayed . . . by a delay circuit 1.” *Id.* ¶¶ 18, 31. Ikeda discloses that reference signal Vr “is generated by obtaining the difference between the output of third microphone unit M3 and the output of delay circuit 1 with inversion adding circuit 4.” *Id.* ¶ 31, Fig. 8. Ikeda further discloses that reference signal Vr is generated by subtracting the output signal of microphone M1, as delayed by delay circuit 1, from the output signal of microphone M3. *Id.* ¶¶ 18, 23, 31, Fig. 8. According to Ikeda, the signal system constructed using microphones M1 and M3, delay circuit 1, and inversion adding circuit 4 for outputting reference signal Vr functions as a reference microphone. *Id.* ¶¶ 12, 20, 23–24, 31, Fig. 4.

Ikeda states that the “reference microphone has directional characteristics on the back surface side.” *Id.* ¶¶ 23–24, Fig. 5. Ikeda explains that the main microphone is configured to pick up a target sound and noise, and the reference microphone is configured to pick up noise. *Id.* ¶¶ 21, 23. The main signal V_m (as converted to a digital signal D_m) and the reference signal V_r (as converted to a digital signal D_r) in Ikeda are input to the adaptive noise cancellation system including an adaptive filter 7 and a digital adder 8. *Id.* ¶¶ 12–13, 27–29, 31, Fig. 1. Ikeda states “[m]ain [sic Reference] signal data D_r outputted from the A/D converter 6 is supplied to an adaptive filter 7 . . . , and the difference between the filter output of the adaptive filter 7 and the main signal data D_m outputted from the A/D converter 5 is calculated by a digital adder 8.” *Id.* ¶ 27, Fig. 1. Ikeda then discloses that the filter response characteristics of adaptive filter 7 “are determined so as to cancel the respective noise components contained in the main signal data D_m and the reference signal data D_r .” *Id.* ¶ 28. Ikeda further discloses

[a]s a result of the response characteristics of the adaptive filter 7 being determined successively in this way, the noise component contained in the main signal V_m is cancelled by the noise component contained in the reference signal V_r due to subtraction by digital adder 8. The output of the digital adder 8 is converted to an analog signal by a D/A converter 9. A signal primarily consisting of the target sound component can be acquired from the D/A converter 9.

Id. ¶¶ 29, 12, 13, 31.

2. *McCowan (Ex. 1007)*

McCowan is an article titled “Near-Field Adaptive Beamformer for Robust Speech Recognition” published in the Digital Signal Processing

Review Journal. Ex. 1007, 87.⁹ McCowan discloses an adaptive noise cancellation system for near-field speech sources with a near-field adaptive beamformer that is “implemented using the standard generalized sidelobe canceler (GSC) system structure.” *Id.* McCowan uses a “near-field compensation unit” and “a standard generalized sidelobe canceling blocking matrix and adaptive filters.” *Id.* at 88–89.

One embodiment of McCowan is shown in Figure 3, reproduced below.

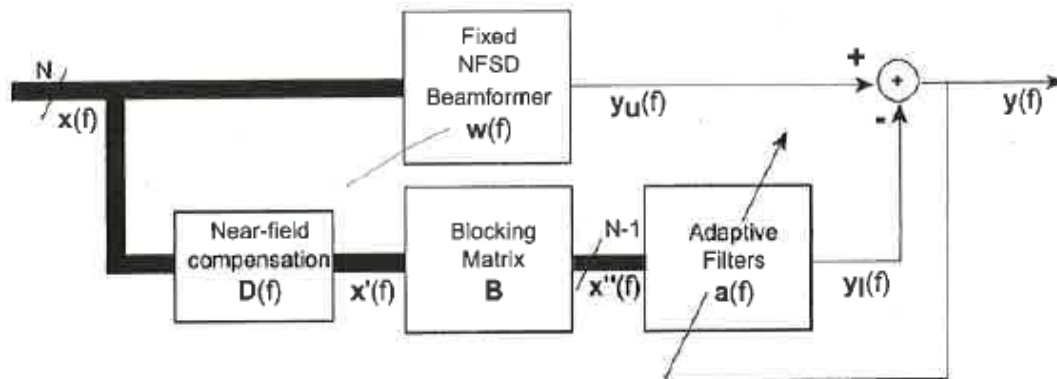


FIG. 3. Near-field adaptive beamformer.

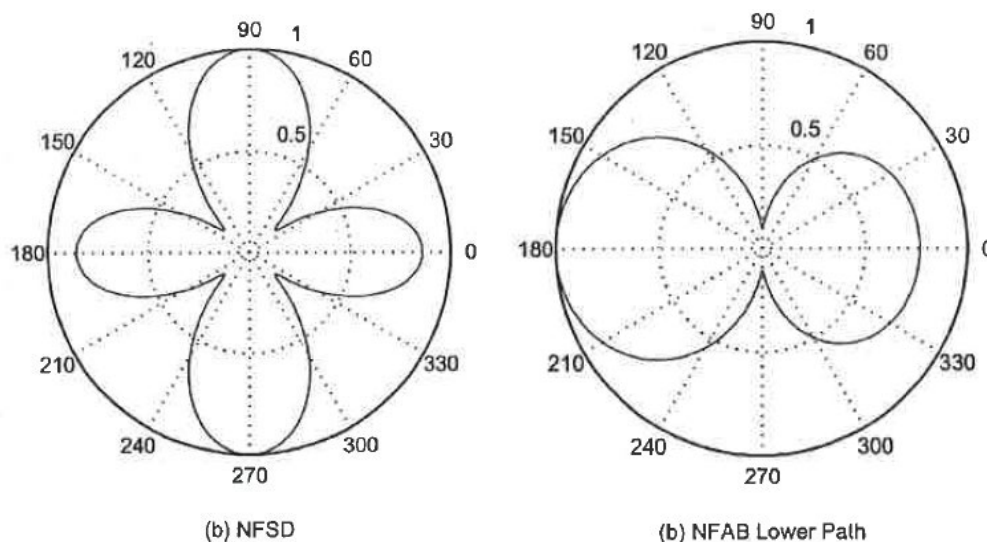
Figure 3, above, illustrates a block diagram of McCowan’s adaptive noise cancellation system with a near-field adaptive beamformer. *Id.* at 91.

McCowan discloses that input $x(f)$ including signals from multiple microphones is processed via two signal paths. *Id.* at 90, 93 (equation 15), 95–96. McCowan’s Figure 3 shows that the lower path includes a near-field compensation, a blocking matrix, and adaptive filters. *Id.* McCowan explains that the lower path generates a noise estimate, which is used to cancel, at a subtraction component, noise in the output signal of a fixed

⁹ To avoid confusion, the page citations are based on the numbers printed in the upper corners of McCowan.

NFSD (near-field superdirective) beamformer in the upper path. *Id.* at 87–91, 93–96.

McCowan’s Figures 6(b) and 7(b) (reproduced below) show that the upper path includes a near-field adaptive beamformer (“NFAB”), that includes at least four nulls, while its lower path NFAB includes at least two nulls, and many more at higher frequencies:



Ex. 1007, 98–99. The “overall beamformer directivity pattern” of McCowan’s NFAB also includes four nulls. *Id.* at 100.

McCowan discloses that the blocking matrix “block[s] the desired signal from the adaptive noise estimate” by subtracting one microphone signal from another. *Id.* at 93, 95 (equations 18, 19). McCowan states that “[t]o ensure complete blocking, the desired signal must both be time aligned and have equal amplitudes across all channels.” *Id.* at 93. McCowan discloses that for a near-field sound source, phase misalignment across microphones results from “[t]he difference in propagation time to [the i th] microphone with respect to the reference microphone ($i=1$),” defined as:

$$\tau_i = \frac{d_i - d_1}{c}$$

where d_i is the distance between the source and the i th microphone and c is the speed of sound. Ex. 1007, 89–90. Additionally, according to McCowan, amplitude differences across microphones are “negligible for far-field sources, but can be significant in the near-field case.” *Id.* at 90. McCowan discloses that the attenuation factor of wave amplitude on the i th microphone, with respect to the amplitude on the reference microphone ($i=1$), is defined as:

$$\alpha_i = \frac{d_1}{d_i}$$

McCowan explains that “[f]or a near-field desired source, to align the desired signal on all channels, a near-field compensation must first be applied to the input channels prior to blocking.” *Id.* at 93; *see id.* at 87. According to McCowan, the near-field compensation unit “compensate[s] for both phase misalignment and amplitude scaling of the desired signal across sensors.” *Id.* at 93. With the near-field compensation, McCowan teaches multiplying microphone 2’s signal by d_2/d_1 , delaying by $(d_1 - d_2)/c$, and subtracting the result from microphone 1’s signal to create “a true null . . . in the beam-pattern . . . in the direction *and* distance corresponding to the desired source.” *Id.* at 90, 93–95.

3. *Kanamori (Ex. 1008)*

Kanamori is a U.S. Patent Application Publication, published on September 23, 2004, titled “Microphone Device and Audio Player.”

Ex. 1008, codes (12), (43), (54). Kanamori discloses a technique “to generate a main signal and noise reference signal used in a noise suppressing process.” *Id.* ¶ 18. One embodiment of Kanamori is shown in Figure 17A, reproduced below.

FIG. 17A

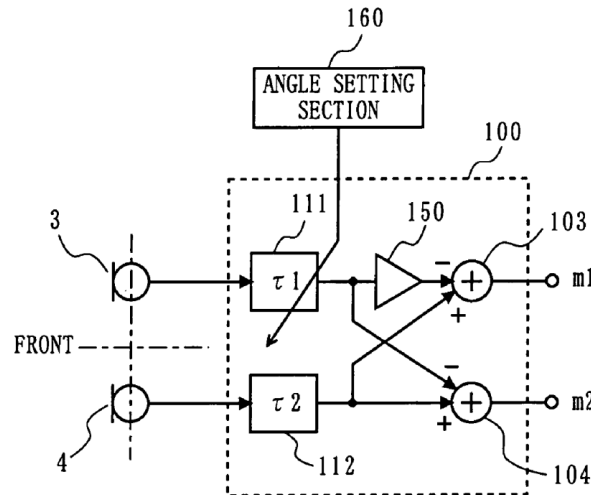


Figure 17A is a schematic perspective view of the configuration of a microphone device according to Kanamori’s embodiment number 8. *Id.* ¶ 66. Specifically, Figure 17A illustrates a technique, where a main signal m1 and a noise reference signal m2 are generated using signals from microphone units 3 and 4. *Id.* ¶¶ 151, 166, 172–174. In this embodiment, directivity combining section **100** includes, in addition to the components of directivity combining section **100**, angle setting section **160** and second signal delaying section **112**. *Id.* ¶ 173. According to Kanamori, the target sound direction can be changed by angle setting section **160**. *Id.* ¶ 174.

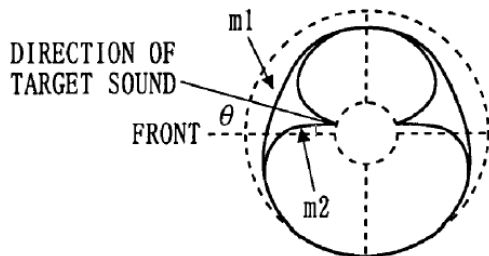
Kanamori discloses that

angle setting section **160** can change a signal delay amount $\tau 1$ 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). Here, if the second signal delaying section **112**

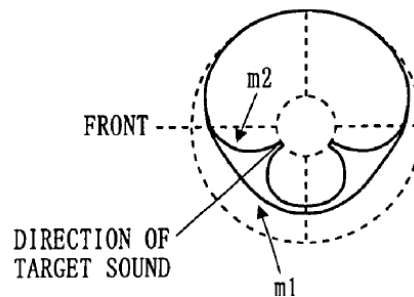
is not provided, even with the signal delay amount τ_1 of the first signal delaying section **111** being changed in the above range, the target sound direction can be changed merely in a range of 0 to +90 degrees with respect to the front direction. With the second signal delaying section **112** being provided and its signal delay amount τ_2 being set as $\tau_2 = d/c$, the target sound direction can be changed in a range of ± 90 degrees with respect to the front direction.

Ex. 1008 ¶ 175. Based on this information, Kanamori states “the direction of collecting sounds (the target sound direction) of the microphone device can be changed.” According to Kanamori, “it is possible to achieve a directivity pattern illustrated in Figure 17B [reproduced below], as well as a directivity pattern illustrated in Figure 17C [reproduced below] by changing the signal delay amount of the signal delay section.” *Id.* ¶ 176.

F I G. 1 7 B



F I G. 1 7 C



Figures 17B and 17C are illustrations showing directivity patterns of Kanamori’s microphone device. *Id.* ¶ 66. Kanamori teaches that the “variable delay characteristic can be easily attained by forming the signal delaying section with an all-pass filter $H(w)=(A+z^{-1})/(a+Az^{-1})$ where a coefficient A is $0 \leq A < 1$.” *Id.* ¶ 176. Kanamori further teaches that “[t]o change the signal delay amount, the angle setting section **160** changes this coefficient A.” *Id.* According to Kanamori,

“[i]f a large delay amount or a linear delay frequency characteristic is required, a secondary all-pass filter and/or an all-pass filter is subordinately connected.” *Id.*

Kanamori discloses “suppress[ing], based on the noise reference signal, noise components included in the main signal.” Ex. 1008 ¶ 168; *see id.* ¶¶ 76–115, Figs. 1–7. Kanamori further discloses a filter coefficient calculating section 40 that calculates the power spectrums of the main and reference signals and, based on the ratio of those power spectrums, calculates the estimated noise spectrum in the main signal, which is used to suppress “noise components included in the main signal.” *Id.* ¶¶ 100–115, 168, Fig. 4; *see id.* ¶¶ 80–86, 93–99.

Kanamori states that, when “noise is coming from a plurality of directions,” suppressing “noise components included in the main signal” requires that the ratio of the power spectrums of the main and reference signals “be constant irrespectively of the direction from which noise is coming.” *Id.* ¶ 168. Otherwise, according to Kanamori, the estimated noise spectrum in the main signal will be correct only “in a specific direction.” *Id.* Kanamori also teaches 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. *Id.* ¶¶ 167–170.

4. Yang (Ex. 1009)

Yang is a U.S. Patent Application Publication, published on December 19, 2002, titled “Noise Suppression for a Wireless Communication Device.” Ex. 1009, codes (12), (43), (54). Yang discloses

an adaptive noise reduction system with “[t]echniques to suppress noise from a signal comprised of speech plus noise.” Ex. 1009, code (57), ¶¶ 58–62. One embodiment of Yang is shown in Figure 4, reproduced below.

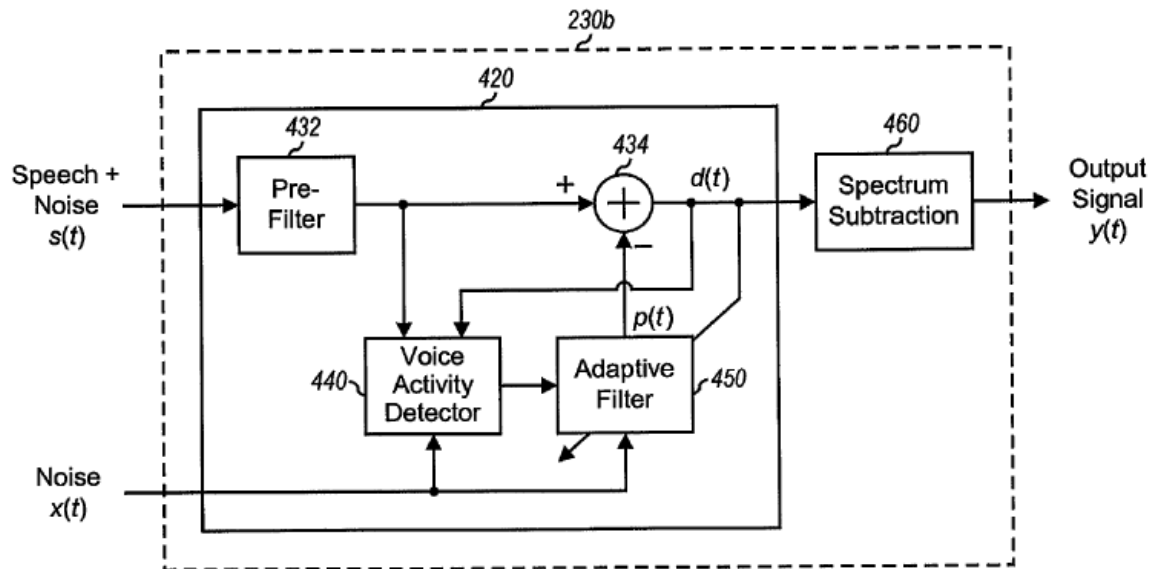


FIG. 4

Figure 4 is a block diagram of Yang’s noise suppression unit 230b. *Id.* ¶¶ 18, 58. Yang discloses that noise suppression unit 230b capable of removing background noise from a speech-plus-noise signal and it achieves the noise reduction/suppression in the time domain. *Id.* ¶ 58.

Yang states:

The mostly noise signal $x(t)$ is provided to an adaptive filter 450, which filters the noise with a particular transfer function $h(t)$. The filtered noise *[sic]* $p(t)$ is then provided to summer 434 and subtracted from the filtered speech plus noise signal to provide an intermediate signal $d(t)$ having predominantly speech and some amount of noise.

Id. ¶ 59. According to Yang, adaptive filter 450 “is adapted during periods of non-speech activity.” *Id.* ¶¶ 60–61. Yang discloses that voice activity detector 440 “detects the presence of speech activity . . . and provides a

control signal that enables the adaptation of the coefficients of [adaptive filter 450] when no speech activity is detected.” *Id.*

IV. ANALYSIS

A. Alleged Obviousness of Claims 1–3, 5–9, 11–14, and 16–20 in view of Ikeda, McCowan, and Kanamori

Petitioner contends claims 1–3, 5–9, 11–14, and 16–20 would have been obvious to person of ordinary skill in the art in view of the combined teachings of Ikeda, McCowan, and Kanamori. Pet. 2, 19–79. Patent Owner disputes Petitioner’s contentions. Prelim. Resp. 1–18. For the reasons discussed below, at this stage of the proceeding, we are persuaded that Petitioner has established a reasonable likelihood of success on this challenge.

1. Analysis of Independent Claim 7

- a) “A method comprising: a first virtual microphone formed from a first combination of a first microphone signal and a second microphone signal, wherein the first microphone signal is generated by a first physical microphone and the second microphone signal is generated by a second physical microphone; a second virtual microphone formed from a second combination of the first microphone signal and the second microphone signal, wherein the second combination is different from the first combination, wherein the first virtual microphone has a first linear response to speech and first linear response to noise, the first linear response to speech being substantially similar across a plurality of frequencies for a speech source located within a predetermined angle relative to an axis of the microphone array and devoid of a null, wherein the second virtual microphone has a second linear response to speech that has a single null oriented in a direction toward a source of the speech and a second linear response to noise, wherein the second linear response to noise is substantially similar to the first linear response to noise, one or both of the first linear response to noise and the second linear response to noise being non-zero in a direction toward a source of*

noise, and the second linear response to speech is substantially dissimilar to the first linear response to speech, wherein the speech is human speech”

Petitioner contends that Ikeda discloses omnidirectional physical microphone units M1 and M2 and the output e1 of microphone M1 is a first microphone signal generated by the first physical microphone, and that output e2 of microphone M2 is a second microphone signal generated by the second physical microphone. Pet. 37–38 (citing Ex. 1006 ¶¶ 18–20, Figs. 1–2; Ex. 1006 ¶ 95; Ex. 1001, 34:16–54). According to Petitioner, Ikeda in combination with McCowan and Kanamori teaches the generation of main signal Vm as well as “a first virtual microphone formed from a first combination of a first microphone signal and a second microphone signal.” *Id.* at 40 (citing Ex. 1006 ¶¶ 18–22, 27; Ex. 1003 ¶¶ 96–98).

Petitioner further contends Ikeda discloses a second virtual microphone by disclosing the reference microphone that constructs reference signal Vr that “[t]he reference microphone has directional characteristics” and includes two omnidirectional microphones M1 and M2 and associated signal processing (e.g., delay circuit 1 and inversion adding circuit 4). *Id.* at 41 (citing Ex. 1006 ¶¶ 18–20, 23–24, Figs. 1, 4–5; Ex. 1003 ¶¶ 99–100). Petitioner argues that “[i]n the reference microphone, a delayed version of output e1 and output e2 are combined (a second combination of the first microphone signal and the second microphone signal) at inversion adding circuit 4, which outputs reference signal Vr.” *Id.* (citing Ex. 1006 ¶¶ 18–20, 23–24; Ex. 1003 ¶¶ 99–100).

Petitioner then contends that a person of ordinary skill in the art would have understood that the combined teachings of Ikeda, McCowan, and Kanamori would have rendered “main signal Vm with linear responses

because the main signal V_m's response to a sound source at a particular distance, frequency, and direction is linearly proportional to its input.” *Id.* at 44 (citing Ex. 1003 ¶ 106). According to Petitioner, “main signal V_m has linear responses just like the virtual microphone V₁ described in the ’080 patent has linear responses, because they are constructed using the same signal processing.” *Id.* at 45 (citing Ex. 1001, 2:60–3:4, 4:42–43, 11:1–2, 11:20–35, Figs. 11–13; Ex. 1003 ¶ 106).

Petitioner relies on the testimony of and a simulation created by Dr. Vipperman to support its position that the linear response of the proposed combination to speech is substantially similar across a plurality of frequencies for a speech source located within an angle of approximately ± 30 degrees relative to an axis of the microphone array. *Id.* at 46 (citing Ex. 1003 ¶¶ 107–108). Thus, Petitioner, argues that in Dr. Vipperman’s simulation “the linear response to speech is devoid of a null.” *Id.* (citing Ex. 1001, 4:31–32 (defining the term “null”); Ex. 1003 ¶¶ 107–108). Again, relying on Dr. Vipperman’s simulation, Petitioner asserts that a person of ordinary skill in the art would have understood that in the combined teachings of Ikeda, McCowan, and Kanamori “reference signal V_r has linear responses because the reference signal V_r’s response to a sound source at a particular distance, frequency, and direction is linearly proportional to its input.” *Id.* at 48–49 (citing Ex. 1003 ¶ 113).

Petitioner also contends that Ikeda discloses applying its teachings “to a wide range of devices,” including “a voice input microphone of a hands-free telephone unit or a car navigation system.” *Id.* at 53 (citing Ex. 1006 ¶ 36). According to Petitioner, a person of ordinary skill in the art would have understood that “voice input” refers to human speech, thereby

rendering it obvious to use Ikeda's system to receive human speech. *Id.* (citing Ex. 1006 ¶ 36; Ex. 1003 ¶ 125).

Patent Owner has not provided any arguments regarding Ikeda and these specific claim limitations. *See generally* Prelim. Resp. Nonetheless, the burden remains on Petitioner to demonstrate unpatentability. *See Dynamic Drinkware*, 800 F.3d at 1378.

At this stage of the proceeding, we are satisfied Petitioner has shown that Ikeda in combination with the teachings of McCowan and Kanamori would have render it obvious to create method comprising

a first virtual microphone formed from a first combination of a first microphone signal and a second microphone signal, wherein the first microphone signal is generated by a first physical microphone and the second microphone signal is generated by a second physical microphone;

a second virtual microphone formed from a second combination of the first microphone signal and the second microphone signal, wherein the second combination is different from the first combination, wherein the first virtual microphone has a first linear response to speech and first linear response to noise, the first linear response to speech being substantially similar across a plurality of frequencies for a speech source located within a predetermined angle relative to an axis of the microphone array and devoid of a null, wherein the second virtual microphone has a second linear response to speech that has a single null oriented in a direction toward a source of the speech and a second linear response to noise, wherein the second linear response to noise is substantially similar to the first linear response to noise, one or both of the first linear response to noise and the second linear response to noise being non-zero in a direction toward a source of noise, and the second linear response to speech is substantially dissimilar to the first linear response to speech, wherein the speech is human speech

as required by claim 7. We credit the testimony of Dr. Vipperman that a simulation based by the proposed Ikeda, McCowan, and Kanamori combination would have created (1) one or both of the first linear response to noise and the second linear response to noise being nonzero in a direction toward a source of noise and (2) a second linear response to speech that would have been substantially dissimilar to a first response to speech. *See* Ex. 1003 ¶¶ 82–93, 119, 122. Consequently, for purposes of institution, we determine Petitioner has shown sufficiently that the proposed Ikeda, McCowan, and Kanamori combination would have rendered the required claim limitations obvious to a person of ordinary skill of the art at the critical time.

- b) *“an adaptive noise removal application coupled to the first and second virtual microphones and generating denoised output signals by forming a plurality of combinations of signals output from the first virtual microphone and the second virtual microphone, by filtering and summing the plurality of combinations of signals in the time domain”*

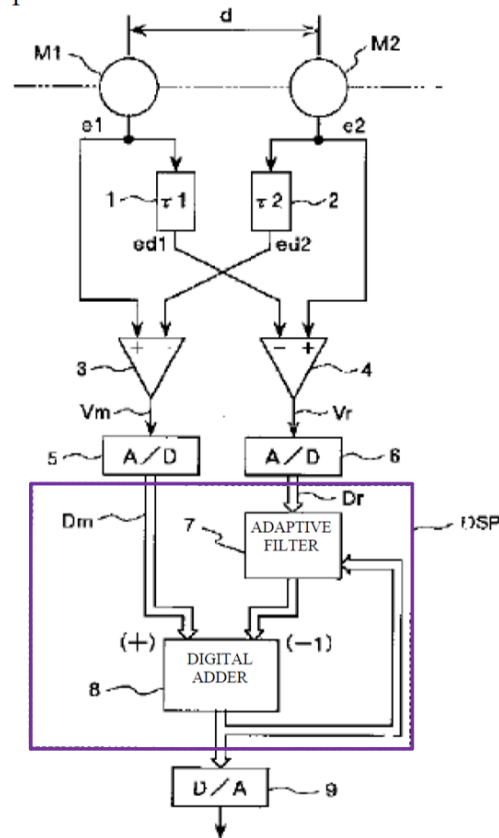
Claim 7 also recites “an adaptive noise removal application coupled to the first and second virtual microphones and generating denoised output signals by forming a plurality of combinations of signals output from the first virtual microphone and the second virtual microphone, by filtering and summing the plurality of combinations of signals in the time domain.”

Ex. 1001, 34:16–54.

Petitioner contends that Ikeda discloses an adaptive noise cancellation system including adaptive filter 7 and digital adder 8, which is implemented using a digital signal processor and therefore is an adaptive noise removal “application.” Pet. 53–54 (citing Ex. 1006 ¶¶ 16, 17, 27–30, Fig. 1; Ex. 1003 ¶ 126). Petitioner specifically argues that “the adaptive noise

cancellation system is coupled to the main and reference microphones and the corresponding main signal V_m and reference signal V_r . . . via A/D converters 5 and 6.” *Id.* at 54 (citing Ex. 1006 ¶¶ 11–13, 18, 31, Figs. 1, 8; Ex. 1003 ¶ 58). Petitioner annotated Figure 1, as reproduced below, with a box around the DSP to indicate what it contends is the adaptive noise cancellation system.

FIG. 1



According to Petitioner, “Ikeda’s adaptive noise cancellation system forms a plurality of combinations of signals output from the first virtual microphone and the second virtual microphone because main signal V_m and reference signal V_r are continuously converted to digital signals and input to the adaptive noise cancellation system implemented on a digital signal processor DSP so that, over time, they form multiple combinations of signals

output from the first and second virtual microphones.” *Id.* at 55 (citing Ex. 1006 ¶¶ 27–29; Ex. 1003 ¶ 129; Ex. 1001, 32:61–63). Petitioner argues its contention is consistent with the ’080 patent’s description that an adaptive noise removal system “combin[es] the two microphone signals (e.g., Mic1, Mic2) by filtering and summing in the time domain.” *Id.* at 56 (citing Ex. 1001, 7:4–16, 8:50–9:3, 9:22–38; Ex. 1003 ¶ 129). Petitioner also asserts that Ikeda’s adaptive noise cancellation system filters and sums the plurality of combinations of signals in the time domain because “Ikeda discloses that digital versions of reference signal V_r are ‘supplied to an adaptive filter’ (filtering) and the output of adaptive filter 7 and digital versions of main signal V_m are differenced at digital adder 8 (summing).” *Id.* at 55–56 (citing Ex. 1006 ¶¶ 27–29; Ex. 1003 ¶ 130).

Petitioner also relies on Dr. Vipperman’s testimony to support its position. Dr. Vipperman opines that person of ordinary skill in the art at the time of the alleged invention would have understood that “adaptive filter 7 filters in the time domain, and digital adder 8 adds in the time domain [because] (1) Ikeda’s system does not include components for converting signals from the time domain to the frequency domain or from the frequency domain to the time domain, and (2) a signal picked up by a microphone is in the time domain if it is not processed by a component and converted to the frequency domain.” Ex. 1003 ¶ 130 (citing Ex. 1006 ¶¶ 27–29; Ex. 1009, 42–43, 58–62).

At this stage of the proceeding, we are satisfied Petitioner has demonstrated sufficiently that Ikeda discloses the challenged limitation and we credit Dr. Vipperman’s supporting testimony. *See* Ex. 1003 ¶ 130. Specifically, we find that Figure 1 of Ikeda includes notations indicating that

adaptive filter 7 and digital adder 8 are part of the DSP (i.e., digital signal processor) and that V_m is combined with V_r in digital adder 8, which then may recirculate together into adaptive filter 7. Thus, we are persuaded that Ikeda teaches a system where V_m does go through adaptive filter 7.

Additionally, if the entire DSP of Ikeda is considered the adaptive noise removal application, then both V_r and V_m go directly into the application.

In view of the evidence presented at this preliminary stage of the proceedings, and for purposes of institution, we determine Petitioner has sufficiently shown that Ikeda teaches “an adaptive noise removal application coupled to the first and second virtual microphones and generating denoised output signals by forming a plurality of combinations of signals output from the first virtual microphone and the second virtual microphone, by filtering and summing the plurality of combinations of signals in the time domain” as required by the challenged claims.

- c) *“by a varying linear transfer function between the plurality of combinations of signals, wherein the denoised output signals include less acoustic noise than acoustic signals received at the first and second physical microphones”*

Claim 7 recites “by a varying linear transfer function between the plurality of combinations of signals, wherein the denoised output signals include less acoustic noise than acoustic signals received at the first and second physical microphones.” Ex. 1001, 34:16–54.

Petitioner contends Ikeda discloses this limitation. Pet. 56–58 (citing Ex. 1003 ¶¶ 131–132). Petitioner specifically contends Ikeda’s adaptive noise cancellation system discloses a varying transfer function between the plurality of combinations of signals because Ikeda discloses that adaptive filter 7’s characteristics “are varied in real time” by “a tap coefficient

updating operation for updating the tap coefficient based on an error signal obtained as the difference between the filter output and a signal from the target system to return the filter output,” which is successively repeated to rewrite the tap coefficient. *Id.* at 56–57 (citing Ex. 1006 ¶¶ 27–29; Ex. 1003 ¶ 131). According to Petitioner, this “error signal is defined as the output of the digital adder 8,” i.e., where the plurality of combinations of signals are summed. *Id.* at 57 (citing Ex. 1006 ¶ 28; Ex. 1003 ¶ 131). Petitioner argues that Ikeda discloses performing this tap coefficient updating operation when “the microphone units M1 and M2 do not pick up the target sound,” which “is the time period in which the output of digital adder 8 should be zero (because there is no speech).” *Id.* (citing Ex. 1006 ¶ 28; Ex. 1003 ¶ 131). Petitioner then argues that by updating the tap coefficient over time to minimize this error signal, the adaptive filter calculates and applies a varying transfer function between the plurality of combinations of signals that results in the output of adaptive filter 7 equaling the digital signal from main signal Vm at a time when there is no speech. *Id.* (citing Ex. 1006 ¶ 28; Ex. 1003 ¶ 131).

Petitioner also contends Ikeda discloses that this transfer function is linear because it discloses that adaptive filter 7 is “a transversal filter” and “perform[s] an output operation for obtaining filter output by multiplying the input signal Dr by a tap coefficient and adding the value in multiple tap units.” *Id.* at 57–58 (citing Ex. 1006, ¶¶ 27–29; Ex. 1003 ¶ 132). According to Dr. Vipperman, a person of ordinary skill in the art at the critical time would have understood that this disclosure in Ikeda teaches a linear filter. Ex. 1003 ¶ 132 (citing Ex. 1020, 16–17, Fig. 2.2 (“Adaptive linear combiner in the form of single-input adaptive transversal filter”); Ex. 1028, 366–367

(“Because of the chain of delay elements across the top of the diagram [of Figure 6.31], this structure is also referred to as a tapped delay line structure or a transversal filter structure. As seen from Figure 6.31, the signal at each tap along this chain is weighted by the appropriate coefficient (impulse response value), and the resulting products are summed to form the output”), Fig. 6.31).

Patent Owner has not provided any arguments regarding Ikeda and this specific claim limitation. *See generally* Prelim. Resp. Nonetheless, the burden remains on Petitioner to demonstrate unpatentability. *See Dynamic Drinkware*, 800 F.3d at 1378.

Based on the current record, we are persuaded by Petitioner’s position. Ikeda discloses that

The adaptive filter 7 is a filter that adaptively varies its characteristics with respect to changes in the input signal over time. That is, the filter characteristics are varied in real time by performing an output operation for obtaining filter output by multiplying the input signal D_r by a tap coefficient and adding the value in multiple tap units, and a tap coefficient updating operation for updating the tap coefficient based on an error signal obtained as the difference between the filter output and a signal from the target system to return the filter output, and successively rewriting the tap coefficient using the updating operation. The error signal is defined as the output of the digital adder 8 at a training sequence timing for determining the filter response characteristics. The training sequence timing is defined as a timing at which the microphone units M1 and M2 do not pick up the target sound. Therefore, the filter response characteristics of the adaptive filter 7 are determined so as to cancel the respective noise components contained in the main signal data D_m and the reference signal data D_r at the training sequence timing.

Ex. 1006 ¶ 28. This appears to be consistent with the '080 patent's disclosure regarding transfer functions $H_1(z)$ and $H_2(z)$. *See* Ex. 1001, 4:52–7:16.

The '080 patent discloses that transfer function $H_1(z)$ “can be calculated using any of the available system identification algorithms and the microphone outputs when the system is certain that only noise is being received.” *Id.* at 5:40–43. The '080 patent explains that the “calculation can be done adaptively, so that the system can react to changes in the noise.” *Id.* at 5:43–44. The '080 patent then explains that the final unknown, function $H_2(z)$, “can be determined by using the instances where speech is being produced and the VAD equals one.” *Id.* at 5:46–48. According the '080 patent, “[i]f the transfer functions $H_1(z)$ and $H_2(z)$ can be described with sufficient accuracy, then the noise can be completely removed and the original signal recovered,” and “[t]his remains true without respect to the amplitude or spectral characteristics of the noise.” *Id.* at 6:22–26. Based on this teaching, we understand that Ikeda is calculating its adaptive filter characteristics in the same manner and for the same reasons the '080 patent calculates transfer functions $H_1(z)$ and $H_2(z)$.

Our understanding is supported by the testimony of Dr. Vipperman, who testifies that “[b]y updating the tap coefficient over time to minimize this error signal, such as by using well known techniques like least mean squares, the adaptive filter calculates and applies a varying transfer function between the plurality of combinations of signals that results in the output of adaptive filter 7 equaling the digital signal from main signal V_m at a time when there is no speech.” *See* Ex. 1003 ¶ 131 (citing Ex. 1006 ¶ 28).

Additionally, we credit Dr. Vipperman’s testimony that a person of ordinary skill in the art at the critical time would have understood that this disclosure in Ikeda teaches a linear filter. *See id.* ¶ 132 (citing Ex. 1020, 16–17, Fig. 2.2; Ex. 1028, 366–367). Dr. Vipperman citation to Ex. 1028 is particularly instructive because the reference explains that a tapped delay line structure, such as that used in Ikeda, is a type of a linear filter. *See* Ex. 1028, 367. Thus, we are persuaded that Ikeda teaches a linear filter, which results in linear proportionality.

Accordingly, at this stage of the proceeding, based on Ikeda itself and the supporting testimony of Dr. Vipperman, we are satisfied Petitioner has shown sufficiently that Ikeda teaches “by a varying linear transfer function between the plurality of combinations of signals, wherein the denoised output signals include less acoustic noise than acoustic signals received at the first and second physical microphones.”

d) Rationale to Combine the Teachings of Ikeda, McCowan, and Kanamori

Petitioner provides numerous other reasons why a person of ordinary skill in the art could and would have combined the teachings of Ikeda and Kanamori with the teachings of McCowan. Pet. 19–34 (citing Ex. 1003 ¶¶ 61–78). Petitioner specifically contends that although “Ikeda assumes far-field speech and noise,” it nonetheless would have been obvious to a person of ordinary skill in the art at the critical time “to implement Ikeda’s system in a device intended to receive near-field speech, such as a headset.” *Id.* at 19 (citing Ex. 1003 ¶ 61).

Petitioner argues that Ikeda discloses applying its invention “to a wide range of devices,” including “a voice input microphone of a hands-free

telephone unit.” *Id.* at 19–20 (citing Ex. 1003 ¶ 61). According to Petitioner, “[i]t was well-known that a headset was such ‘a voice input microphone of a hands-free telephone unit.’” *Id.* at 20 (citing Ex. 1006 ¶ 36; Ex. 1011 ¶ 4; Ex. 1012 ¶¶ 8–9; Ex. 1013 ¶¶ 48, 71, 126, Abstract; Ex. 1003 ¶ 61). Petitioner asserts that “[s]uch a headset was commonly located 3–5” (approximately 0.08–0.13 meters) away from the user’s mouth” and that “the speech source is in the near-field and one or more noise sources are in the far-field.” *Id.* (citing Ex. 1012 ¶¶ 9, 78; Ex. 1003 ¶ 61; Ex. 1007; Ex. 1011 ¶¶ 6, 35).

Petitioner further argues that when “Ikeda’s system is implemented in a device intended for environments where the speech source is in the near-field and one or more noise sources are in the far-field, such as a headset, a [person of ordinary skill in the art] would have been motivated to improve Ikeda’s system in view of McCowan.” *Id.* at 20–21 (citing Ex. 1003 ¶¶ 62–66). Petitioner specifically asserts that a person of ordinary skill in the art “would have been motivated to modify the Ikeda reference microphone to add the gain β_1 in view of McCowan” resulting in a signal system where “the gain and the delay minimizes leakage of a desired signal of a near-field source into the noise reference signal.” *Id.* at 23 (citing Ex. 1003 ¶¶ 62–66), 23 (citing Ex. 1007, 87–91, 93–96, Fig. 3; Ex. 1003 ¶ 64).

Petitioner then argues that a person of ordinary skill in the art would have had a reasonable expectation of success to make modifications and combine the teachings of Ikeda and McCowan with that of Kanamori due to the similarities between Ikeda and Kanamori. *Id.* at 33 (citing Ex. 1003 ¶ 78). Additionally, Petitioner asserts that “both Ikeda and Kanamori use signals from two microphones to form a signal system for generating a main

signal having directional characteristics, and this similarity would reinforce the reasonable expectation of success to make the modification.” *Id.* (citing Ex. 1006 ¶¶ 18–22, Figs. 1–3; Ex. 1008 ¶¶ 163–176, Figs. 17A–17C; Ex. 1003 ¶ 78). According to Petitioner, “[m]odifying the signal path with a delay and a gain would have been within the knowledge and skill of a [person of ordinary skill in the art].” *Id.* at 34 (citing Ex. 1003 ¶ 78; Ex. 1007, 90, 93–95; Ex. 1008 ¶ 163–176). Petitioner, therefore, concludes that such modifications “would have resulted in predictable results that the reference microphone outputs a noise reference signal where leakage of a desired signal of the near-field speech source into the noise reference signal is minimized.” *Id.* at 25–26 (citing Ex. 1006 ¶¶ 18–20, 23–24, Figs. 1, 4–5; Ex. 1007, 87–91, 93–96, Fig. 3; Ex. 1003 ¶ 66).

Patent Owner contends that Petitioner fails to identify legitimate motivations to combine the teachings of the proffered references to arrive at the claimed invention. Prelim. Resp. 21–22. First, Patent Owner argues that the Petition fails to “articulate a fact-based motivation to implement Ikeda in a headset.” *Id.* at 22. According to Patent Owner, simply “recognizing that there are specific near-field applications of stated far-field applications” is insufficient to provide a reason to make any specific near-field application. *Id.* at 22–23. Patent Owner further asserts that Petitioner fails to explain why a person of ordinary skill in the would have chosen to make a headset among the myriad of unidentified applications that Ikeda’s system could have supported, much less those which would have required modifications to Ikeda. *Id.* at 23 (citing Ex. 2002 ¶ 67). Patent Owner concludes that Petitioner’s arguments are premised on impermissible hindsight analysis of the challenged claims. *Id.*

Second, Patent Owner argues that the Petition fails to provide any evidence or an explanation as to why a person of ordinary skill in art would have been motivated to combine Ikeda with McCowan to make a headset or to add a gain function to the Ikeda circuit in view of the teachings of McCowan. Prelim. Resp. 23. Patent Owner also argues that McCowan does not disclose a headset, so (1) a person of ordinary skill in art would not have looked to McCowan to provide a component to modify Ikeda to make a headset, and (2) Petitioner does not explain why a person of ordinary skill in art would have looked to McCowan to modify Ikeda to make a headset. *Id.* at 25.

Patent Owner next argues that Petitioner cherry-picks a single gain from McCowan’s circuit in order to modify Ikeda and make it appear closer to the claimed invention.” Prelim. Resp. 25. According to Patent Owner, “if McCowan is a near-field reference and Ikeda is not,” a person of ordinary skill in art also would have modified both Ikeda’s Vm and Vr circuits according McCowan’s disclosure, but the Petition only modifies Ikeda’s Vr circuit but not Ikeda’s Vm circuit and fails to explain why it only modifies the single circuit. *Id.* at 25–28 (citing Ex. 2002 ¶¶ 72–74).

Third, Patent Owner contends the Petition fails to address why a person of ordinary skill in the art 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.” Prelim. Resp. 29. Patent Owner asserts that McCowan was chosen for combination with Ikeda because it discloses a near-field application. *Id.* (citing Pet. 22, 23). But, according to Patent Owner, Petitioner uses 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*." *Id.* Patent Owner argues that these positions are inconsistent and undermine one another because "either McCowan's status as a near-field device is irrelevant and does not support the combination, or Kanamori's status as a far-field device undermines a motivation to combine to arrive at a near-field device." *Id.* (citing Ex. 2002 ¶ 77).

Patent Owner also contends that the Petition is deficient because (1) it relies on multiple distinct embodiments within Kanamori but fails to identify any motivation to combine the different embodiments, and (2) its proposed modifications to Kanamori removes the inventive aspect shown in Figure 17A. *Id.* at 30–37.

We have reviewed the parties' arguments, the prior art of record, and Dr. Vipperman's testimony regarding the alleged rationale to combine the teachings of Ikeda, McCowan, and Kanamori. At this stage of the proceeding and based on the current record, we are persuaded that Petitioner's showing is adequate to support institution. We understand that Ikeda and Kanamori are directed to far-field systems, while McCowan employs a near-field model. In one instance, however, McCowan discloses having a speaker situated 60 cm from a center microphone, directly in front of an array with a localized noise source at a distance of 270 cm from the array. *See* Ex. 1007, 96. These distances appear to overlap with the far-field distances taught in Ikeda, and the location of the array appears to fall within the scope of Ikeda's "hands-free telephone unit" that is expected to receive noise from every direction. *See* Ex. 1006 ¶ 36.

Additionally, we are persuaded at this stage that Petitioner has shown adequately why a person of ordinary skill in the art would have combined McCowan’s teachings to Ikeda, “so that a null exists in [Ikeda’s] reference microphone’s directivity pattern in the ‘direction *and* distance’ corresponding to the near-field speech source.” *See* Pet. 24–25; Ex. 1007, 94; Ex. 1003 ¶ 65. We also are persuaded based on current information that a person of ordinary skill in the would have been motivated to apply the gain β to the signal of microphone M1 for forming the reference microphone to compensate for the amplitude difference between microphones M1 and M2, as taught by McCowan. *See* Pet. 24–25; Ex. 1007, 89–90, 93–96; Ex. 1003, ¶ 65.

Moreover, we understand Petitioner is relying on Kanamori’s teachings for certain filtering, directivity patterns (Pet. 28–33), and signal delay/signal processing (*id.* at 69 (citing Ex. 1008 ¶ 112), 73 (citing Ex. 1008 ¶ 151), 77 (citing Ex. 1008 ¶ 151), 80). We do not understand that Petitioner is relying on wholesale incorporation of certain Kanamori’s embodiments or specific inventive aspects into Ikeda and/or McCowan. Therefore, we are not persuaded by Patent Owner’s argument, because in our analysis we look to whether the combined teachings of the references would have suggested to one of ordinary skill in the art the challenged claim. “The test for obviousness is not whether the features of a secondary reference may be bodily incorporated into the structure of the primary reference Rather, the test is what the combined teachings of the references would have suggested to those of ordinary skill in the art.” *In re Keller*, 642 F.2d 413, 425 (CCPA 1981) (citations omitted). Accordingly, at

this stage of the proceeding, we are persuaded that Petitioner's showing is sufficient.

2. *Analysis of Claims 1–3, 5, 6, 8, 9, 11–14, and 16–20*

Claims 1 and 14 are independent claims having similar limitations to claim 7, while claims 2, 3, 5, 6, 8, 9, 11–14, and 16–19 depend directly or indirectly from independent claims 1, 7, or 14. Ex. 1001, 33:39–36:36.

Petitioner has provided additional explanation for the manner by which the combination of Ikeda, McCowan, and Kanamori is asserted to disclose each limitation not found in claim 7. Pet. 59–79 (citing Ex. 1003 ¶¶ 136–199).

Patent Owner has not provided any arguments regarding Ikeda, McCowan, and Kanamori in regards to these challenged claims. *See generally* Prelim. Resp. Nonetheless, the burden remains on Petitioner to demonstrate unpatentability. *See Dynamic Drinkware*, 800 F.3d at 1378.

Having reviewed Petitioner's arguments and supporting evidence in this present record, including the arguments summarized above for claim 1, we determine Petitioner has established adequately for purposes of this Decision that the Ikeda discloses the limitations of claims 1–3, 5, 6, 8, 9, 11–14, and 16–20. Accordingly, we determine Petitioner has demonstrated a reasonable likelihood that these claims would have been rendered obvious to a person of ordinary skill in the art by the combined teachings of Ikeda, McCowan, and Kanamori.

B. *Alleged Obviousness of Claims 4, 10, and 15 in view of Ikeda, McCowan, Kanamori, and Yang*

Petitioner contends claims 4, 10, and 15 would have been obvious to person of ordinary skill in the art in view of the combined teachings of Ikeda, McCowan, Kanamori, and Yang. Pet. 2, 79–82 (citing Ex. 1003

¶¶ 202–210). Claims 4, 10, and 15 recite limitations directed to voice activity signals. Ex. 1001, 34:6–8, 34:63–65, 36:12–14. Petitioner argues that Yang discloses using a voice activity detector. Pet. 79–82 (citing Ex. 1009 ¶¶ 58–62). According to Petitioner, it would have been obvious to modify the teachings of Ikeda, McCowan, and Kanamori to include use of a voice activity detector to determine the timing of when microphone do or do not pick up human speech signals because such modifications would have been within the knowledge and skill of a person of ordinary skill in the art at the critical time. *Id.* at 81 (citing Ex. 1003 ¶ 205).

Patent Owner disputes Petitioner’s contentions for the same reasons it disputes Petitioner’s challenged based on Ikeda, McCowan, and Kanamori. Prelim. Resp. 37. Based on the reasons discussed previously, at this stage of the proceeding, we are persuaded that Petitioner has established a reasonable likelihood of success on this challenge.

V. CONCLUSION

For the foregoing reasons, we determine Petitioner has demonstrated there is a reasonable likelihood it would prevail in establishing the unpatentability of all claims challenged in the Petition. Additionally, we decline to exercise our discretion under 35 U.S.C. § 314 to deny the Petition.

Our factual findings, conclusions of law, and determinations at this stage of the proceeding are preliminary and based on the evidentiary record developed thus far. At this stage of the proceeding, the Board has not made a final determination as to the patentability of any challenged claim. Our final decision will be based on the record as fully developed during trial.

VI. ORDER

In consideration of the foregoing, it is hereby:

ORDERED that, pursuant to 35 U.S.C. § 314(a), an *inter partes* review is hereby instituted as to claims 1–20 of the '080 patent on all challenges presented in the Petition, namely:

- (1) Claims 1–3, 5–9, 11–14, and 16–20 under 35 U.S.C. § 103 as unpatentable over Ikeda, McCowan, and Kanamori;
- (2) Claims 4, 10, and 15 under 35 U.S.C. § 103 as unpatentable over Ikeda, McCowan, Kanamori, and Yang;

FURTHER ORDERED that pursuant to 35 U.S.C. § 314(c) and 37 C.F.R. § 42.4, notice is hereby given of the institution of a trial, the trial commencing on the entry date of this Decision.

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