

Filed: February 3, 2023

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UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

AMAZON.COM, INC.,
Petitioner,

v.

JAWBONE INNOVATIONS, LLC,
Patent Owner.

IPR2023-00559
U.S. Patent No. 8,503,691

**PETITION FOR *INTER PARTES* REVIEW
OF CLAIMS 1-22 AND 29-40 OF U.S. PATENT NO. 8,503,691**

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Exhibit	Description
Ex. 1001	U.S. Patent No. 8,503,691 (“the ’691 patent”)
Ex. 1002	Prosecution History of the ’691 Patent
Ex. 1003	Declaration of Jeffrey S. Vipperman, Ph.D.
Ex. 1004	Curriculum Vitae of Jeffrey S. Vipperman, Ph.D.
Ex. 1005	U.S. Patent Publication No. 2004/0185804 to Kanamori et al. (“Kanamori”)
Ex. 1006	Iain A. McCowan et al., <i>Near-field Adaptive Beamformer for Robust Speech Recognition</i> , 12 Digital Signal Processing 87-106 (2002) (“McCowan”)
Ex. 1007	U.S. Patent No. 5,473,701 to Cezanne et al. (“Cezanne”)
Ex. 1008	U.S. Patent Publication No. 2007/0244698 to Dugger et al. (“Dugger”)
Ex. 1009	Certified Translation of Japanese Unexamined Patent Application Publication No. H11-18186 (“Ikeda”)
Ex. 1010	U.S. Patent Application Publication No. 2008/0152167 to Taenzer (“Taenzer”)
Ex. 1011	U.S. Patent Application Publication No. 2006/0120537 to Burnett et al. (“Burnett”)
Ex. 1012	U.S. Patent Application Publication No. 2003/0031328 to Elko et al. (“Elko ’328”)

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Exhibit	Description
Ex. 1013	U.S. Patent No. 8,098,844 to Elko (“Elko ’844”)
Ex. 1014	[Placeholder to match exhibit numbering in IPR2022-01060]
Ex. 1015	Excerpts from Lawrence E. Kinsler et al., <i>Fundamentals of Acoustics</i> , John Wiley & Sons, Inc. (4th ed. 2000)
Ex. 1016	Excerpts from M. P. Norton et al., <i>Fundamentals of Noise and Vibration Analysis for Engineers</i> , Cambridge Univ. Press (2d ed. 2003)
Ex. 1017	Declaration of Shauna L. Wiest Regarding McCowan
Ex. 1018	[Placeholder to match exhibit numbering in IPR2022-01060]
Ex. 1019	U.S. Patent No. 8,942,387 to Elko et al. (“Elko ’387”)

Petitioner Amazon.com, Inc., requests *inter partes* review of claims 1-22 and 29-40 of U.S. Patent 8,503,691, which Jawbone Innovations, LLC (“Patent Owner” or “PO”), purportedly owns.

I. PRECISE RELIEF REQUESTED

Petitioner requests cancellation of claims 1-22 and 29-40 of U.S. Patent No. 8,503,691 (Ex. 1001) as unpatentable based on the following:

Prior art:

Ex. 1005	U.S. Patent Publication No. 2004/0185804 to Kanamori et al. (“Kanamori”), filed November 18, 2003, and published September 23, 2004; prior art to the ’691 patent under pre-AIA 35 U.S.C. §§102(a)-(b) and (e).
Ex. 1006	Iain A. McCowan et al., <i>Near-field Adaptive Beamformer for Robust Speech Recognition</i> , 12 Digital Signal Processing 87-106 (2002) (“McCowan”); published in January 2002; prior art to the ’691 patent under pre-AIA 35 U.S.C. §§102(a)-(b).

Unpatentability ground:

Ground 1	Kanamori and McCowan render obvious claims 1-22 and 29-40.
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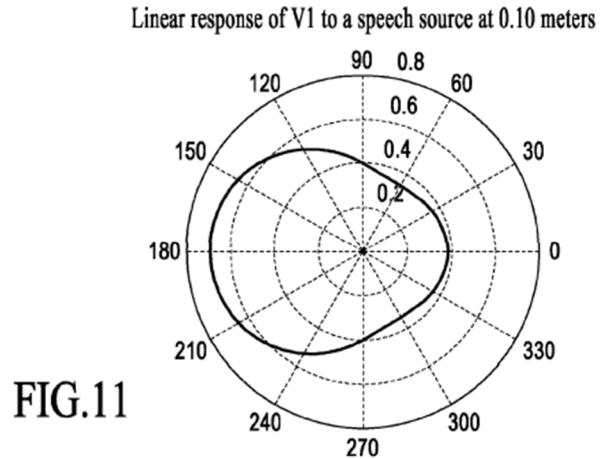
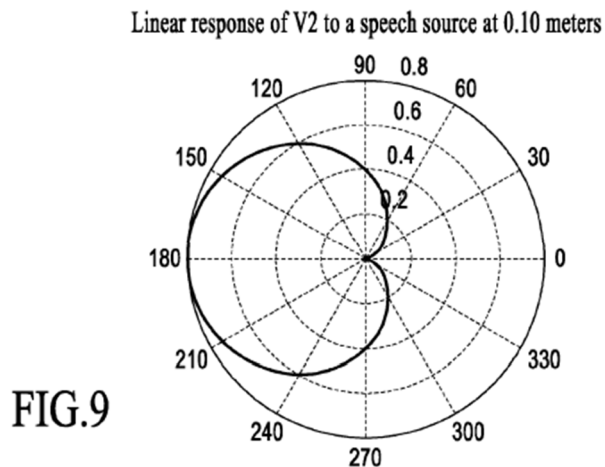
II. '691 PATENT

A. Specification

The '691 patent discloses generating “virtual microphones” from a pair of physical microphones that “are configured to have very similar noise responses and very dissimilar speech responses.” *Id.*, Abstract, 4:63-67. Virtual microphones include those “constructed using two or more omnidirectional microphones and associated signal processing.” *Id.*, 5:59-62. The '691 patent discloses forming “a wide variety of virtual microphones” by “varying the magnitude and sign of the delays and gains of the processing paths.” *Id.*, 9:5-8.

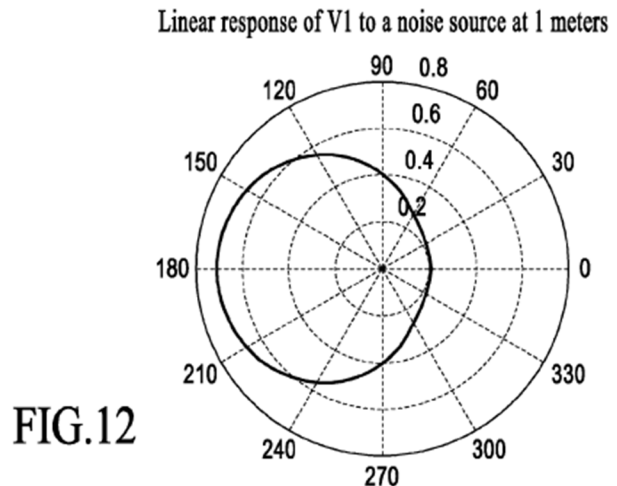
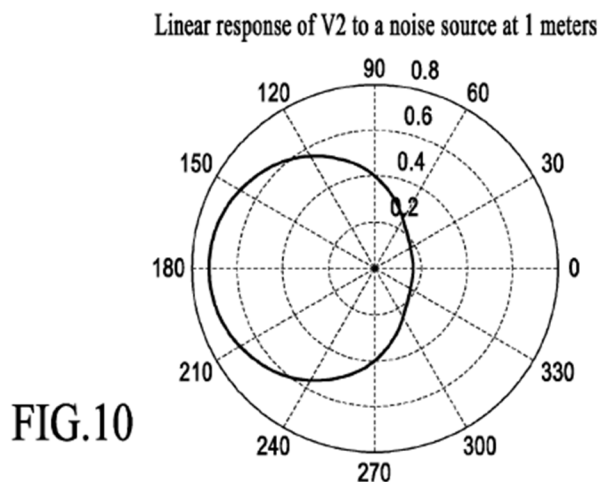
The '691 patent discloses a first virtual microphone that is a “virtual directional ‘speech’ microphone” and “has no nulls,” and a second virtual microphone that is a “virtual directional ‘noise’ microphone” and “has a null for the user’s speech.” *Id.*, 5:53-56. The virtual microphones are constructed so that the second virtual microphone “includes relatively small speech response,” the first virtual microphone “includes sufficient speech response,” and the first and second virtual microphones include “substantially similar noise response.” *Id.*, 10:26-42.

Figures 9 and 11 depict the linear responses of the second and first virtual microphones, respectively, to a speech source. *Id.*, 11:21-23, 12:20-22, Figs. 9, 11. In these figures, a near-field speech source located at 0° and 0.1 m from the microphone array is assumed. *Id.*, 12:61-64, 17:5-7.



Id., Figs. 9, 11.

Figures 10 and 12 depict the linear responses of the second and first virtual microphones, respectively, to a noise source. Ex. 1001, 11:25-28, 12:25-27, Figs. 10, 12. In these figures, a far-field noise source located 1.0 meter from the microphone array at all angles is assumed. *Id.*, 12:61-64, 17:5-7.



Id., Figs. 10, 12.

B. Prosecution History

The PTO Examiner did not consider the references Petitioner relies on. In fact, despite the generic nature of the claims and wealth of available prior art, the PTO Examiner cited only two references against the issued claims. Ex. 1001. The applicant did not submit any references in an IDS. *See id.* After the applicant amended the independent claims multiple times in response to rejections based on the same two references, the Examiner allowed the claims. Ex. 1002, 132-137, 164-167, 178-187, 190-193, 203-221, 235-236. In the notice of allowance, the “Reasons for Allowance” merely state: “a dual virtual microphone array as claimed including having a linear response to both speech and noise could not be anticipated nor, in combination, be rendered obvious over the prior art of record.” *Id.*, 235-236. The applicant never argued that linear responses to both speech and noise were important to patentability.

C. Level of Ordinary Skill in the Art

Based on the relevant factors, *In re GPAC Inc.*, 57 F.3d 1573, 1579 (Fed. Cir. 1995), a person of ordinary skill in the art (“POSITA”) would have had a minimum of 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. Ex. 1003, ¶25. Work experience can substitute for formal education and additional formal education can substitute for work experience. *Id.*

D. Claim Construction

Only disputed terms need to be construed, and only to the extent necessary to resolve the dispute. *Nidec Motor Corp. v. Zhongshan Broad Ocean Motor Co.*, 868 F.3d 1013, 1017 (Fed. Cir. 2017). To the extent the '691 patent arguably defines certain terms, *see* Ex. 1001, 5:21-62, those meanings are discussed in this Petition, and construing the terms is not necessary. As discussed below, the prior art is substantially the same as the '691 patent's disclosed embodiments and, therefore, renders the claims unpatentable without claim construction.

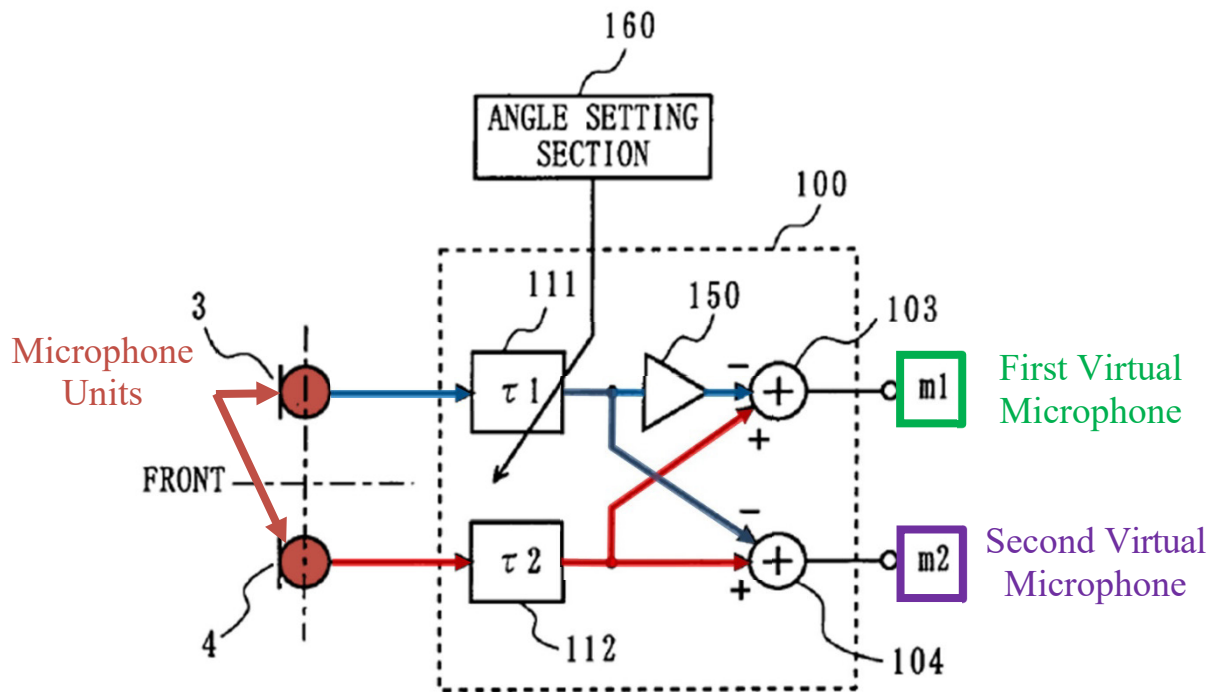
III. CLAIMS 1-22 AND 29-40 WOULD HAVE BEEN OBVIOUS OVER KANAMORI AND MCCOWAN.

A. Overview of Kanamori

Kanamori discloses microphone devices that “use a plurality of microphone units having non-directivity . . . to obtain directivity by combining signals output from these microphone units.” Ex. 1005, ¶[0151]. Such devices require the microphones “to be a certain distance (normally, 1 cm to 5 cm) apart,” provide “flexibility in designing directivity,” and allow the use of “a variable characteristic using a digital process.” *Id.*, ¶[0151]. Kanamori uses such a directivity-combining process “to

generate a **main signal** and **noise reference signal**,” which are used to detect “a desired sound coming from a specific direction with noise being suppressed.” *Id.*, ¶¶[0002], [0018], [0172]-[0173]. Figure 17A depicts one such directivity-combining process that constructs **main signal m1** and **noise reference signal m2** using **microphone units 3 and 4** and associated signal processing. *Id.*, ¶¶[0172]-[0173].

F I G. 1 7 A



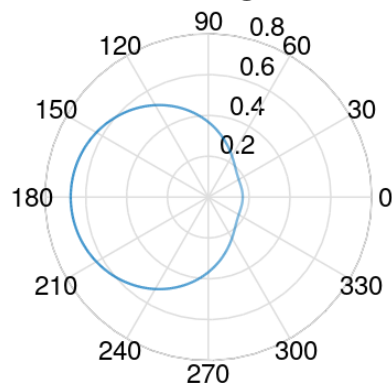
Id., Fig. 17A (annotated).

m1's and **m2**'s directivity patterns are constructed such that “the directions of a minimum sensitivity in the directivity are in the direction of the target sound” so that **m1** and **m2** include more noise components “than components of the target

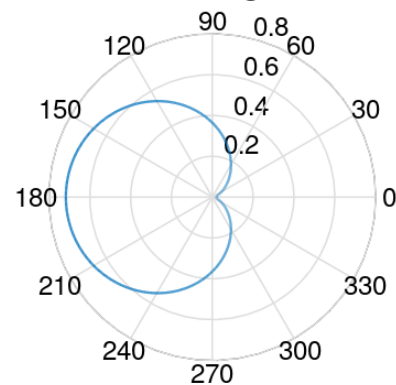
sound.” *Id.*, ¶[0167]. The shape of their directivity patterns is preferably different “from each other only at the portions of directions of a minimum sensitivity” and is “identical” everywhere else. *Id.*, ¶[0168]. Kanamori states that this helps “to suppress, based on the noise reference signal, noise components included in the **main signal** under the circumstances where noise is coming from a plurality of directions.” *Id.*, ¶[0168].

Kanamori’s **m1** and **m2** from Figure 17A have “a large difference in sensitivity” in “the direction of the target sound” and have substantially no difference in sensitivity to sound sources from all other directions when the sound source originates in the far field. *Id.*, ¶[0168]; Ex. 1003, ¶¶52-53. For example, when the target sound source is expected from 0°, plots of the linear responses of **m1** and **m2** to a 1000 Hz sound source 1 m from the microphone array show that there is a large difference in sensitivity around the direction of the target sound source and substantially no difference in sensitivity in all other directions. Ex. 1003, ¶¶52-53.

m1, Kanamori Fig. 17a at 1m



m2, Kanamori Fig. 17a at 1m



Ex. 1003, ¶52.

B. Overview of McCowan

McCowan teaches a technique for improving noise cancellation when a speech source is in the near field using **near-field compensation** in the signal path for generating a noise reference signal via a **blocking matrix**, as depicted in Figure 3, below. Ex. 1006, 87, 90-91; Ex. 1003, ¶¶44-48.

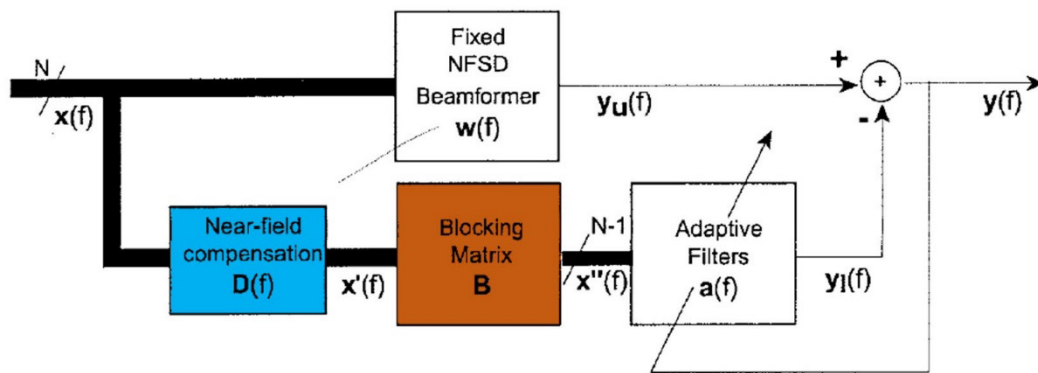


FIG. 3. Near-field adaptive beamformer.

Id., Fig. 3 (annotated).

According to McCowan, “[m]any microphone array processing techniques assume a planar signal wavefront,” which “is reasonable for a far-field source,” as in Kanamori. *Id.*, 89. But for a near-field speech source, “a more accurate spherical wavefront model must be employed.” *Id.*, 89. McCowan discloses two issues that [near-field compensation](#) must account for: “phase misalignment” and “amplitude scaling.” *Id.*, 89-90, 93.

First, “the differences in distance to each sensor can be significant for a near-field source, resulting in phase misalignment across sensors.” *Id.*, 89. To align the phase, McCowan discloses the following equation representing “[t]he difference in propagation time to each microphone with respect to the reference microphone ($i=1$)”:

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

where d_i represents the distance from the source to the i th microphone and c is the speed of sound. *Id.*, 89-90.

Second, “the wavefront amplitude decays at a rate proportional to the distance traveled.” *Id.*, 90. “The resulting amplitude differences across sensors are negligible for far-field sources, but can be significant in the near-field case.” *Id.*, 90. McCowan discloses a microphone-attenuation factor α_i :

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

where d_i is the same as above. *Id.*, 90.

Combining both phase misalignment and amplitude scaling, McCowan discloses that “if $x_1(f)$ is the desired source at the reference microphone, the signal on the i th microphone is given by

$$x_i(f) = \alpha_i x_1(f) e^{-j2\pi f \tau_i}.”$$

Id., 90. McCowan’s **near-field compensation** requires delaying a reference signal x_1 by the difference in propagation time τ_i between x_1 and x_i and multiplying by α_i , the distance from the source to the reference microphone divided by the distance from the source to the i th microphone. *Id.*, 89-90; Ex. 1003, ¶48. Performing this **near-field compensation** allows a **blocking matrix** to completely “block the desired signal from the adaptive noise estimate” by effectively ensuring that “a true null exists in the beam-pattern of each **blocking matrix** row in the direction *and* distance corresponding to the desired source.” *Id.*, 91, 93-94.

McCowan is prior art to the ’691 patent because it was publicly accessible to those skilled in the art in 2002 when it published in *Digital Signal Processing*, Volume 12, Number 1, a regularly published periodical journal by Elsevier Science. Ex. 1006, Cover, Title Pages, 87; Ex. 1017, ¶¶12-31.

McCowan was received and stamped by the Library of Congress on or before February 19, 2002, and would have been processed, cataloged, shelved, and made

available to the public a few days or weeks after receipt. Ex. 1017, ¶¶12-17, 23-27, Appx. A, B, C.

McCowan was received and stamped by the library of Iowa State University of Science and Technology on or before February 18, 2002, and would have been processed, cataloged, shelved, and made available to the public a few days or weeks after receipt. Ex. 1017, ¶¶12, 18-22, 28-30, Appx. D, E, F.

McCowan bears a copyright date of 2002 and indicates that it was published in “Digital Signal Processing 12, 87-106 (2002).” Ex. 1006, Cover, Title Pages, 87. Because *Digital Signal Processing* was a regular periodic journal that was distributed widely and intended to be accessible to the public, McCowan’s copyright date and publication information are evidence of public availability at that date. *See Microsoft Corp. v. IPA Techs. Inc.*, IPR2019-00811, Paper 44 at 142 (P.T.A.B. Oct. 15, 2020); *Ericsson Inc. v. Intellectual Ventures I LLC*, IPR2014-00527, Paper 41 at 10-12 (P.T.A.B. May 18, 2015).

C. Independent Claim 1

Kanamori generically discloses devices with similar responses to noise and dissimilar responses to speech, but operates when both the speech source and the noise source(s) are in the far field and does not differentiate based on distance of the sound source to the microphones. But devices such as headsets that receive near-field speech were well known before the ’691 patent, and McCowan teaches how to

modify Kanamori's second virtual microphone to make its noise-cancellation technique effective for such devices, rendering obvious claim 1.

1. [1P]: "A microphone array comprising:"

Kanamori discloses or suggests [1P].

Kanamori discloses a "microphone device" (e.g., an audio or video recorder) that includes two microphone units. Ex. 1005, ¶¶[0077], [0151], [0201], Figs. 14-21, 23. The two microphone units constitute a microphone array. Ex. 1003, ¶62.

2. [1A]: "a first virtual microphone comprising a first combination of a first microphone signal and a second microphone signal,"

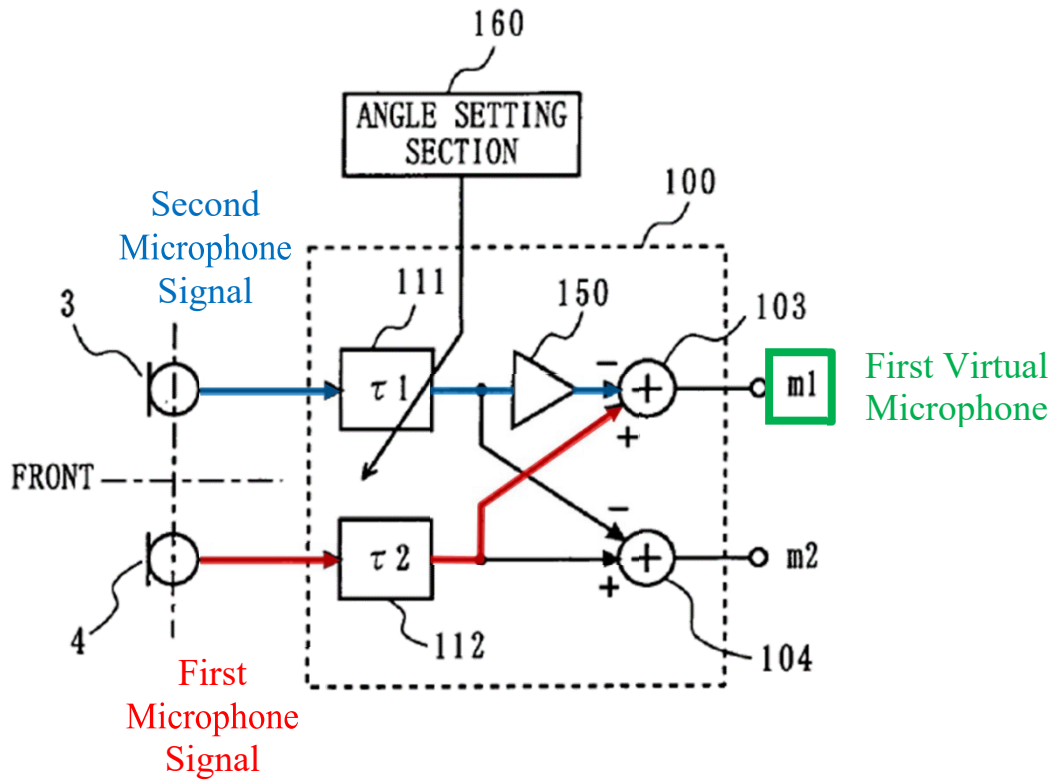
Kanamori discloses or renders obvious [1A].

The '691 patent states that a "virtual microphone[]" is "a microphone constructed using two or more omnidirectional microphones and associated signal processing." Ex. 1001, 5:59-62.

Kanamori's **main signal m1** is a first virtual microphone. Ex. 1003, ¶63. Kanamori discloses that devices "often use a plurality of microphone units having non-directivity," i.e., two or more omnidirectional microphones, "to obtain directivity by combining signals output from these microphone units," i.e., associated signal processing. Ex. 1005, ¶[0151]; Ex. 1003, ¶63. In Figure 17A, **main signal m1** comprises a combination of "**a signal output from the fourth microphone unit 4**" and "**a**

signal output from the third microphone unit 3,” corresponding to the first and second microphone signals, respectively. Ex. 1005, ¶¶[0166], [0172], [0174]; Ex. 1003, ¶63.

FIG. 17A



Ex. 1005, Fig. 17A (annotated).

Kanamori discloses signal processing associated with constructing m1. Ex. 1003, ¶64. As depicted above, a “signal output from the fourth microphone unit 4” is delayed by “the second signal delaying section 112.” Ex. 1005, ¶¶[0166], [0174]. A “signal output from the third microphone unit 3” is delayed by the “signal delaying

section 111” and multiplied by an amplification ratio at the “signal amplifying section 150.” Ex. 1005, ¶¶[0166], [0169], [0174]. A “first signal subtracting section 103” subtracts the **signal output from the signal amplifying section 150** from the **signal output from the second signal delaying section 112** to obtain **m1**. Ex. 1005, ¶¶[0166], [0174].

In Figure 17A, “a plurality of microphone units having the same directivity characteristic are used to obtain the **main signal**.” Ex. 1005, ¶[0172]. A POSITA would understand that Figure 17A discloses using two omnidirectional microphones as the microphone units. Ex. 1003, ¶65. Omnidirectional microphones are examples of microphones having the same directivity characteristic in all directions. Ex. 1003, ¶65; *see* Ex. 1005, ¶[0193] (disclosing two “non-directional microphone units of the same characteristic”), ¶[0025] (describing a directivity-combining section similar to Figure 17A, where the microphones have an “identical” characteristic). Kanamori discloses that the microphone units can be either omnidirectional or directional microphones and accordingly discloses constructing **main signal m1** using two omnidirectional microphones. Ex. 1005, ¶¶[0020] (disclosing obtaining the main signal “by performing a predetermined process on a signal output from *any microphone unit* (that is, a non-directional microphone unit or a directional microphone unit whose main axis of directivity is oriented to a predetermined direction).” (emphasis

added)), [0113] (disclosing using a “unidirectional” or “non-directional” microphone), [0151] (stating that devices can use “a plurality of microphone units having non-directivity or directivity of the same characteristic to obtain directivity by combining signals output from these microphone units”), [0193] (disclosing an embodiment with two “non-directional microphone units of the same characteristic”); Ex. 1003, ¶65.

Moreover, it would have at least been obvious to use omnidirectional microphones as the microphone units in Figure 17A. Ex. 1003, ¶66. Kanamori teaches forming directional microphones by combining signals from multiple omnidirectional microphones. Ex. 1005, ¶¶[0020], [0113], [0151], [0193]. A POSITA would have been motivated to use omnidirectional microphones to avoid “manufacturing variations in characteristics, such as the orientation of the direction of a minimum sensitivity in the directivity and the amount of sensitivity attenuation.” Ex. 1005, ¶[0083]; Ex. 1003, ¶66. Combining signals from omnidirectional microphones to form virtual microphones was well known, giving a POSITA a reasonable expectation of success in using omnidirectional microphones for the microphone units in Figure 17A. Ex. 1003, ¶66; *e.g.*, Ex. 1005, ¶¶[0151], [0193]; Ex. 1008 (Dugger), ¶[0036]; Ex. 1009 (Ikeda), ¶(0011).

3. [1B]: “the first virtual microphone having a first linear response to speech and a first linear response to noise,”

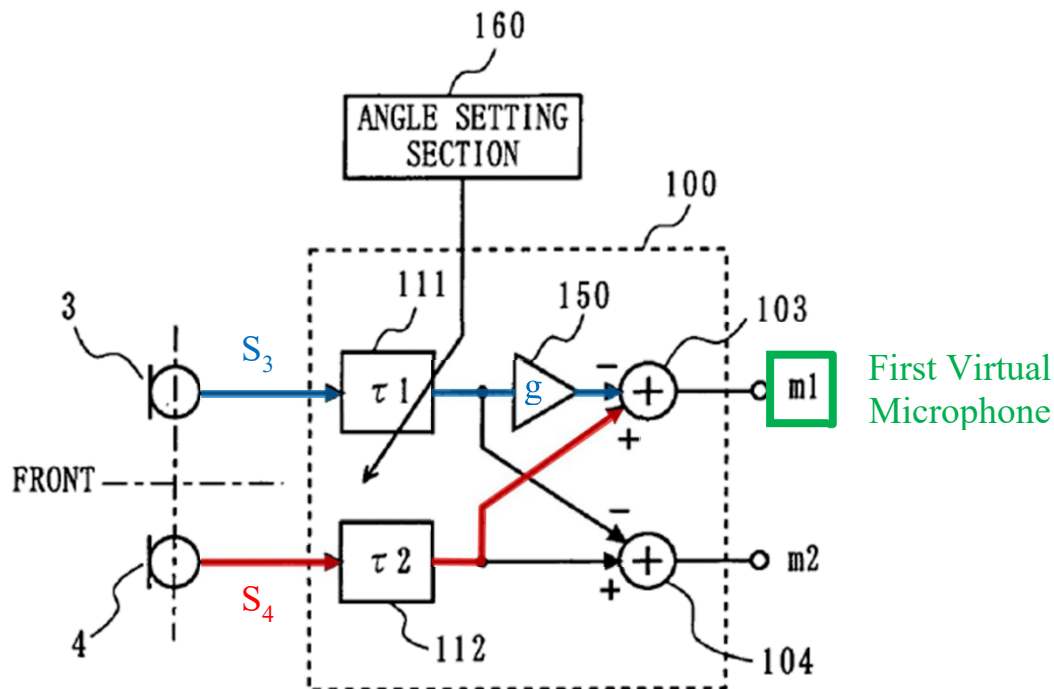
Kanamori discloses or renders obvious [1B].

Kanamori discloses obtaining main signal m1 by subtracting a delayed and amplified version of the signal output from the third microphone unit from a delayed version of the signal output from the fourth microphone unit. Ex. 1005, ¶¶[0166], [0174]; *supra* §III.C.2 ([1A]). Kanamori discloses that this directivity-combining process can be performed digitally. Ex. 1005, ¶¶[0151], [0182]. A POSITA would have understood that m1 has linear responses to speech and noise that can be simulated and visualized, as explained by Dr. Vipperman. Ex. 1003, ¶67. The equation for m1 is provided below in the digital domain:

$$\text{Equation (1): } m1 = z^{-\tau2 \cdot fs} \cdot S_4 - g \cdot z^{-\tau1 \cdot fs} \cdot S_3,$$

where $z^{-\tau2 \cdot fs}$ refers to a delay of time $\tau2$, $z^{-\tau1 \cdot fs}$ refers to a delay of time $\tau1$, g is the gain applied by signal amplifying section 150, S_3 is the “signal output from the third microphone unit 3,” and S_4 is the “signal output from the fourth microphone unit 4.” Ex. 1005, ¶¶[0166], [0174]-[0175]; Ex. 1003, ¶67; *infra* §III.D.8 (claim 12, explaining that z^{-s} refers to a delay of s samples). This equation is further illustrated by Figure 17A:

FIG. 17A



Ex. 1005, Fig. 17A (annotated). Kanamori provides further details regarding the variables and their ranges, including that $\tau 1$ is set to zero and $\tau 2$ is set to d/c when speech is expected to approach from an angle of 0° , as described further below, and the gain is “set at approximately 0.85.” Ex. 1005, ¶¶[0025], [0167], [0169], [0175]; Ex. 1003, ¶67.

The '691 patent discloses plots of linear responses of virtual microphones, where a virtual microphone's linear response to speech is depicted as the directivity plot of the virtual microphone's near-field response, and a virtual microphone's linear response to noise is depicted as the directivity plot of the virtual microphone's far-field response. Ex. 1001, Figs. 9-13, 11:21-29, 12:20-33, 12:61-13:5, 17:5-7

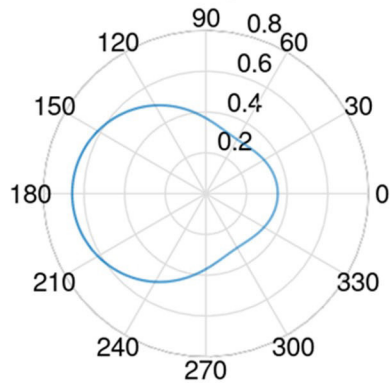
(“[T]he noise (far-field) responses of V_1 and V_2 are very similar while the speech (near-field) responses are very different.”); Ex. 1003, ¶68. The ’691 patent assumes that “the speech is located at approximately 0 degrees and approximately 10 cm,” and “the noise at all angles is located approximately 1.0 meter away from the mid-point of the array.” Ex. 1001, 12:61-64. Kanamori does not explicitly describe **main signal m1** as having different linear responses, depending on the distance of the sound source to the array, but a POSITA would have understood from Kanamori’s description of **main signal m1**, including Equation (1) above, that it has a first linear response to near-field speech and a first linear response to far-field noise. Ex. 1003, ¶68.

Kanamori discloses generic “microphone devices” that “detect[] a desired sound coming from a specific direction with noise being suppressed.” Ex. 1005, ¶[0002]. It would have been obvious to implement Kanamori’s microphone device in a device intended to receive near-field speech, such as a headset. Ex. 1003, ¶69. Kanamori discloses various “use environments” and “purposes” for its microphone devices, including calling and voice recognition. Ex. 1005, ¶¶[0115], [0182], [0207]. A headset was one such conventional microphone device used for calling. Ex. 1008 (Dugger), ¶[0004]; Ex. 1009 (Ikeda), ¶(0036); Ex. 1010 (Taenzer), ¶[0008]-[0009]; Ex. 1011 (Burnett), Abstract, ¶¶[0048], [0071], [0126]; Ex. 1003, ¶69. Such headsets commonly utilized omnidirectional microphones between 3-5”

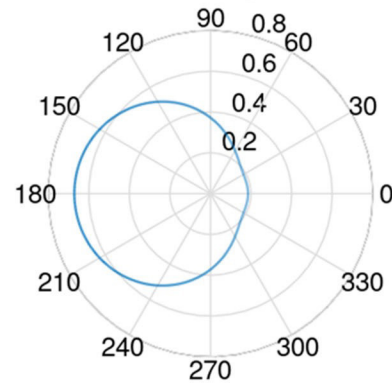
(.08-.13 m) from the user’s mouth. Ex. 1010 (Taenzer), ¶¶[0009], [0078]; Ex. 1011 (Burnett), ¶[0048]; Ex. 1003, ¶69. In these headsets, the microphones are close to a near-field speech source, but noise signals “generally arriv[e] from distant locations” and “simultaneously exist in various directions.” Ex. 1005, ¶[0012]; Ex. 1010 (Taenzer), ¶[0009]; Ex. 1003, ¶69; *see also* Ex. 1006, 88 (disclosing “near-field speech sources”), 89 (disclosing that “a planar signal wavefront” is a reasonable assumption “for a far-field source,” but a “spherical wavefront model must be employed” “when the desired source is close to the array”); Ex. 1008 (Dugger), ¶¶[0006], [0035].

When implemented in a device that receives near-field speech, Kanamori’s **main signal m1** in Figure 17A has a first linear response to near-field speech and a first linear response to far-field noise. Ex. 1003, ¶70. Dr. Vipperman simulated and plotted the linear responses of **main signal m1** to a 1000 Hz signal located 0.1 m (speech) from the microphone array and a 1000 Hz signal located 1 m (noise) from the microphone array when $\tau_1=0$:

m1, Kanamori Fig. 17a at 0.1m



m1, Kanamori Fig. 17a at 1m



Ex. 1003, ¶70 (linear response to speech (left) and linear response to noise (right)). These plots are the same types of plots that the '691 patent calls the “linear response[s]” of the virtual microphone. *See* Ex. 1001, Figs. 9-13, 2:41-60, 11:21-29, 12:20-33; Ex. 1003, ¶70. Additionally, the equations representing m1 from Kanamori and the first virtual microphone in the '691 patent are substantially similar. *Compare* Equation (1), *with* Ex. 1001, 11:67. Moreover, a POSITA would understand that these responses are linear because m1's response to a sound source at a particular distance, frequency, and direction is linearly proportional to its input. Ex. 1003, ¶70. For example, as depicted in the above plots, m1's response to a 1000 Hz speech source at 0.1 m and 0° can be determined by multiplying the input amplitude by about 0.35, and m1's response to a 1000 Hz noise source at 1 m and 90° can be determined by multiplying the input amplitude by about 0.38. Ex. 1003, ¶70.

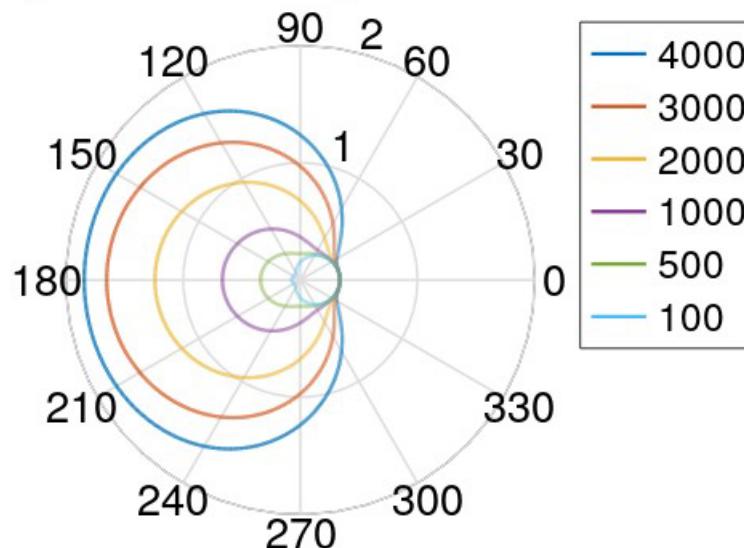
Accordingly, Kanamori's **main signal m1** discloses a virtual microphone having a first linear response to speech and a first linear response to noise. Ex. 1003, ¶71.

4. [1C]: “the first linear response to speech being substantially similar across a plurality of frequencies for a speech source located at an angle relative to an axis of the microphone array,”

Kanamori discloses or suggests [1C].

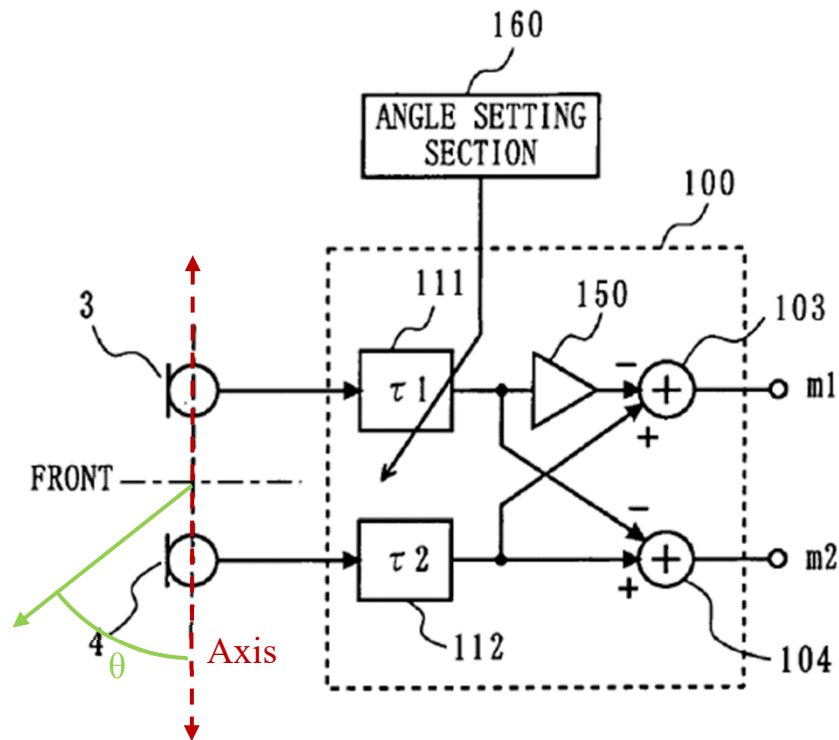
As explained for [1B], when implemented in a device that receives near-field speech, Kanamori's **main signal m1** in Figure 17A has a first linear response to near-field speech. *Supra* §III.C.3; Ex. 1003, ¶72. Dr. Vipperman simulated and plotted the linear response of **main signal m1** to a signal containing various frequencies located 0.1 m (speech) from the microphone array when $\tau_1=0$:

m1, Kanamori Fig. 17a at 0.1m



Ex. 1003, ¶72. In the depiction of **main signal m1**'s linear responses, the angle corresponds to the value of θ in the figure below. Ex. 1003, ¶72.

FIG. 17A



Ex. 1005, Fig. 17A (annotated).

Kanamori discloses that the direction of minimum sensitivity points towards the direction of speech, which corresponds to $\theta=0^\circ$ in the above figure, as desired for $\tau_1=0$. Ex. 1005, ¶¶[0167], [0175]; Ex. 1003, ¶72. As depicted, **m1**'s linear response to speech is substantially similar across a plurality of frequencies for between about ± 30 degrees relative to an **axis** of the microphone array, which includes the direction of speech. Ex. 1003, ¶72. In other words, for this direction range, **m1**'s

response to a speech source between 100 and 4000 Hz at 0.1 m can be determined by multiplying the input amplitude by about 0.35. Ex. 1003, ¶72.

Accordingly, Kanamori's **main signal m1** discloses a virtual microphone having a first linear response to speech that is substantially similar across a plurality of frequencies for a speech source located at an **angle** of 0° relative to an **axis** of the microphone array. Ex. 1003, ¶73.

5. **[1D]: “wherein the first microphone signal is generated by a first physical microphone and the second microphone signal is generated by a second physical microphone; and”**

Kanamori discloses or suggests [1D].

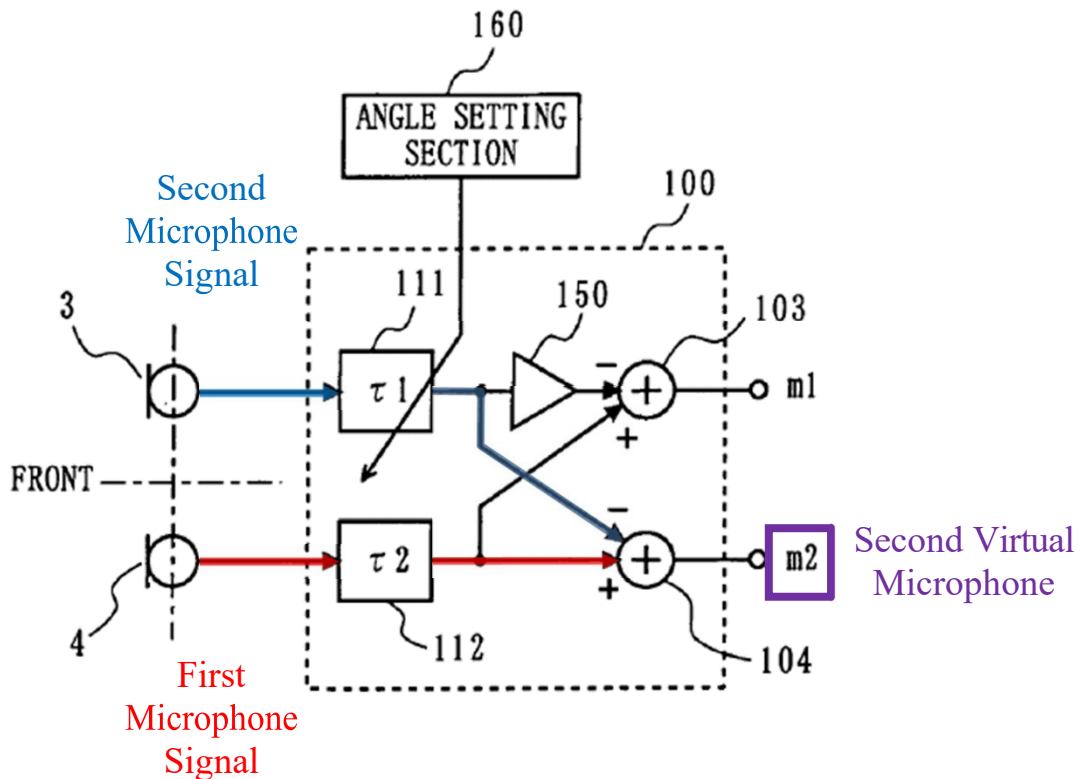
Kanamori discloses a “signal output from the fourth microphone unit 4” (first microphone signal) and a “signal output from the third microphone unit 3” (second microphone signal). Ex. 1005, ¶¶[0166], [0174]. Microphone unit 4 is a first physical microphone that generates the corresponding signal output, and microphone unit 3 is a second physical microphone that generates the corresponding signal output. Ex. 1005, ¶[0151] (disclosing that the microphone units are physical microphones); Ex. 1003, ¶74.

6. **[1E]: “a second virtual microphone comprising a second combination of the first microphone signal and the second microphone signal,”**

Kanamori discloses or renders obvious [1E].

Kanamori's **noise reference signal m2** is a second virtual microphone. Ex. 1003, ¶75. As explained *supra* Section III.C.2 ([1A]), Kanamori discloses or renders obvious that microphone units 3 and 4 in Figure 17A can be omnidirectional microphones. Ex. 1005, ¶¶[0020], [0113], [0151], [0193]; Ex. 1003, ¶¶75, 77. In Figure 17A, **noise reference signal m2** comprises a combination of “**a signal output from the fourth microphone unit 4**” (first microphone signal) and “**a signal output from the third microphone unit 3**” (second microphone signal). Ex. 1005, ¶¶[0166], [0172], [0174]; Ex. 1003, ¶75.

F I G. 1 7 A



Ex. 1005, Fig. 17A (annotated).

Kanamori discloses signal processing associated with constructing **m2**. Ex. 1003, ¶76. As depicted above, a “**signal output from the fourth microphone unit 4**” is delayed by “the second signal delaying section 112.” Ex. 1005, ¶¶[0166], [0174]. A “**signal output from the third microphone unit 3**” is delayed by the “signal delaying section 111.” Ex. 1005, ¶¶[0166], [0174]. A “second signal subtracting section 104” subtracts the **signal output from the signal delaying section 111** from the **signal output from the second signal delaying section 112** to obtain **m2**. Ex. 1005, ¶¶[0166], [0174].

7. [1F]: “the second virtual microphone having a second linear response to speech and a second linear response to noise,”

Kanamori discloses or suggests [1F].

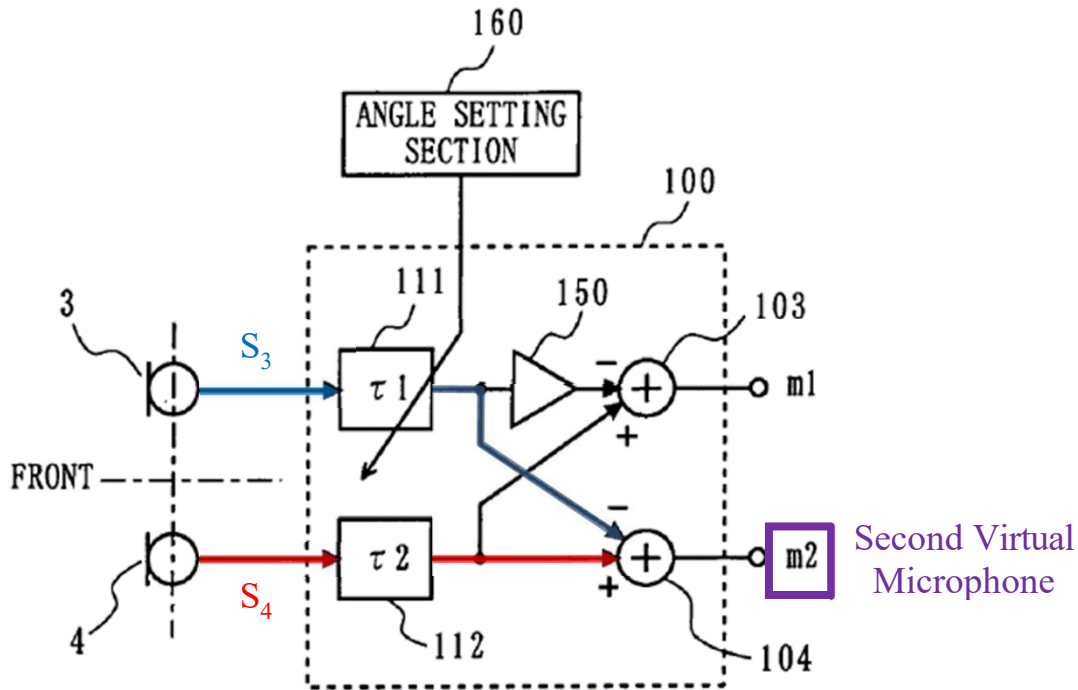
As described for [1E], **noise reference signal m2** is obtained by subtracting a delayed version of the **signal output from the third microphone unit 3** from a delayed version of the **signal output from the fourth microphone unit 4**. Ex. 1005, ¶¶[0166], [0174]; *supra* §III.C.6. Kanamori discloses that this directivity-combining process can be performed digitally. Ex. 1005, ¶¶[0151], [0182]. A POSITA would have understood that **m2** has linear responses to speech and noise that can be simulated and visualized, as explained by Dr. Vipperman. Ex. 1003, ¶78. The equation for **m2** is provided below in the digital domain:

$$\text{Equation (2): } \mathbf{m2} = z^{-\tau_2 \cdot f_s} \cdot \mathbf{S_4} - z^{-\tau_1 \cdot f_s} \cdot \mathbf{S_3},$$

where $z^{-\tau_1 f_s}$, $z^{-\tau_2 f_s}$, S_3 , and S_4 mean the same thing here as they do for Equation (1).

Supra §III.C.3 ([1B]); Ex. 1005, ¶¶[0166], [0174]-[0175]; Ex. 1003, ¶78. This equation is further illustrated by Figure 17A:

F I G. 1 7 A



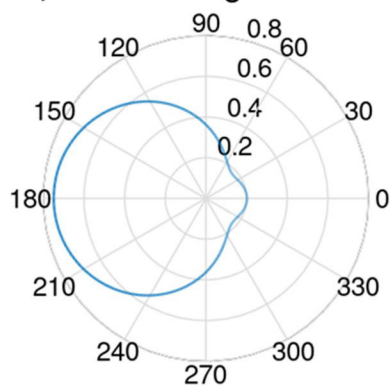
Ex. 1005, Fig. 17A (annotated).

The '691 patent discloses that the linear response to speech refers to the virtual microphone's near-field response, and the linear response to noise refers to the virtual microphone's far-field response. *Supra* §III.C.3 ([1B]). Kanamori does not explicitly describe noise reference signal $m2$ as having different linear responses, depending on the distance of the sound source to the array, but a POSITA would

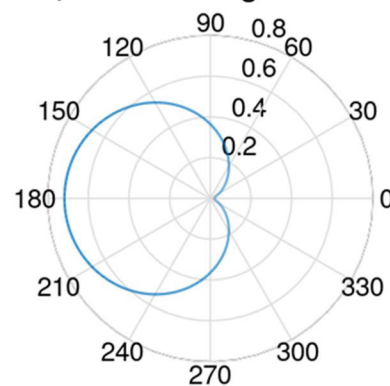
have understood from Kanamori's description of **noise reference signal m2**, including Equation (2) above, that it has a second linear response to near-field speech and a second linear response to far-field noise. Ex. 1003, ¶79.

It would have been obvious to implement Kanamori's microphone device in a device intended to receive near-field speech, such as a headset. *Supra* §III.C.3 ([1B]); Ex. 1003, ¶80. When implemented in such a device, Kanamori's **noise reference signal m2** in Figure 17A has a second linear response to near-field speech and a second linear response to far-field noise. Ex. 1003, ¶80. Dr. Vipperman simulated and plotted the linear responses of **noise reference signal m2** to a 1000 Hz signal located 0.1 m (speech) from the microphone array and a 1000 Hz signal located 1 m (noise) from the microphone array when $\tau_1=0$:

m2, Kanamori Fig. 17a at 0.1m



m2, Kanamori Fig. 17a at 1m



Ex. 1003, ¶80 (linear response to speech (left) and linear response to noise (right)). These plots are the same types of plots that the '691 patent calls the "linear response[s]" of the virtual microphone. See Ex. 1001, Figs. 9-13, 2:41-60, 11:21-29,

12:20-33; Ex. 1003, ¶80. Moreover, a POSITA would understand that these responses are linear because m2's response to a sound source at a particular distance, frequency, and direction is linearly proportional to its input. Ex. 1003, ¶80. For example, as depicted in the above plots, m2's response to a 1000 Hz speech source at 0.1 m and 0° can be determined by multiplying the input amplitude by about 0.2, and m2's response to a 1000 Hz noise source at 1 m and 90° can be determined by multiplying the input amplitude by about 0.38. Ex. 1003, ¶80. As depicted, the direction of minimum sensitivity for both linear responses is towards the source of speech at 0°, as desired for $\tau_1=0$. Ex. 1003, ¶80; Ex. 1005, ¶¶[0167], [0175].

Accordingly, Kanamori's noise reference signal m2 discloses a virtual microphone having a second linear response to speech and a second linear response to noise. Ex. 1003, ¶81.

8. [1G]: “the second linear response to noise being substantially similar to the first linear response to noise, and the second linear response to speech being substantially dissimilar to the first linear response to speech,”

Kanamori and McCowan render obvious [1G].

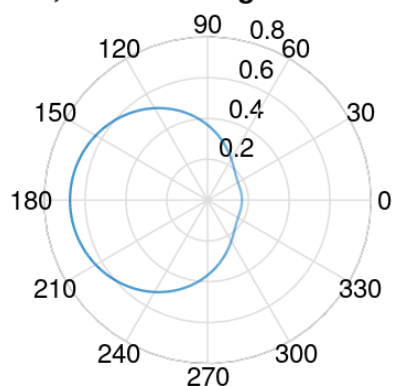
It would have been obvious to modify Kanamori's Figure 17A to implement McCowan's near-field compensation for when the speech source is in the near field. By implementing this near-field compensation, the modified Figure 17A discloses

substantially similar linear responses to noise and substantially dissimilar linear responses to speech.

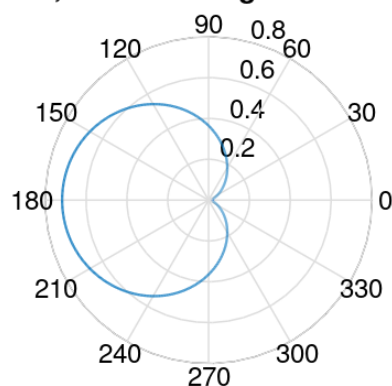
Kanamori discloses a variety of “use environments” and “purposes” for its microphone devices, many of which would not expect speech to always be in the near field. Ex. 1005, ¶[0182]; Ex. 1003, ¶82. For example, Kanamori discloses implementing its microphone device in a “video recorder, capable of collecting sounds.” Ex. 1005, ¶[0151]; Ex. 1003, ¶82. As such, Kanamori’s directivity-combining process for forming m1 and m2 does not explicitly describe factoring in the distance of the speech source to the microphones because the amplitude differences at each microphone caused by amplitude decay from the distance traveled “are negligible for far-field sources.” Ex. 1006, 90; Ex. 1012 (Elko ’328), ¶[0021]; Ex. 1003, ¶82.

For such devices that assume far-field speech, Kanamori discloses that the directivity patterns of m1 and m2 preferably show “a large difference in sensitivity” in “the direction of the target sound” and “no difference” in sensitivity in all other directions. Ex. 1005, ¶[0168]; Ex. 1003, ¶83. This is depicted in Dr. Vipperman’s plots of the linear responses of main signal m1 and noise reference signal m2 to a 1000 Hz signal originating 1 m from the microphone array, assuming a target-sound direction of 0°.

m1, Kanamori Fig. 17a at 1m



m2, Kanamori Fig. 17a at 1m



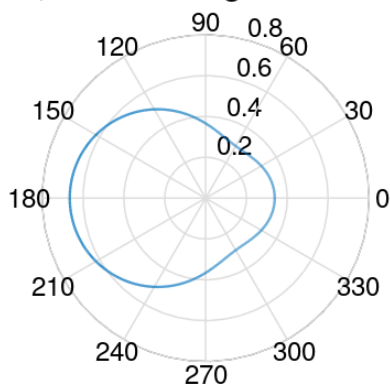
Ex. 1003, ¶83. Kanamori discloses constructing **m1** and **m2** this way because, “to suppress, based on the **noise reference signal**, noise components included in the **main signal** under the circumstances where noise is coming from a plurality of directions,” the sensitivity to noise in the directivity patterns of **m1** and **m2** must be the same “irrespective of the direction from which noise is coming.” Ex. 1005, ¶[0168]; Ex. 1003, ¶83.

Accordingly, when both speech and noise originate in the far field, a POSITA would understand that Kanamori’s microphone device would suppress noise from most directions well but would struggle to suppress noise that originated from the same direction as the speech source because it could not differentiate between speech and noise arriving from the same direction. Ex. 1005, ¶[0168]; Ex. 1003, ¶83.

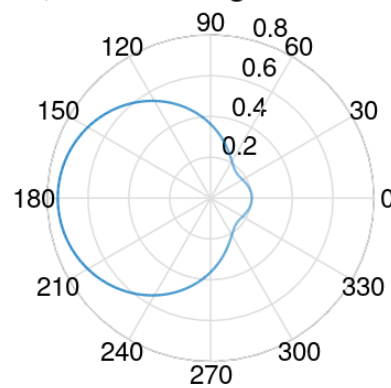
It would have been obvious to implement Kanamori’s microphone device in a device intended to receive near-field speech, such as a headset. *Supra* §III.C.3 ([1B]); Ex. 1003, ¶84. In such a context (e.g., headset), it would have been obvious

to modify Kanamori’s noise reference signal m2 to account for the differences in amplitude decay caused by distance traveled, which “can be significant in the near-field case.” Ex. 1006, 90; Ex. 1003, ¶84. The need for such a modification would have been apparent to a POSITA because, unlike the far-field situation, Kanamori’s noise reference signal m2 is relatively sensitive to near-field speech, as depicted in Dr. Vipperman’s plots of m1’s and m2’s linear responses to speech:

m1, Kanamori Fig. 17a at 0.1m



m2, Kanamori Fig. 17a at 0.1m



Ex. 1003, ¶84. Because “noise is suppressed mainly through a directivity combining process,” a POSITA would have recognized that noise reference signal m2 would make for a poor noise reference signal at this distance. Ex. 1003, ¶84; Ex. 1005, ¶¶[0083], [0182]. Kanamori discloses that, if the target sound enters m2 (as here), “detrimental effects, such as distortion or reduction in level, will occur in the processed sound.” Ex. 1005, ¶[0083]. To avoid such detrimental effects, a POSITA

would have looked to McCowan to determine how to modify Kanamori’s **noise reference signal m2** for use in a device such as a headset. Ex. 1003, ¶84.

McCowan recognizes the above problem, stating that “leakage of the desired signal” is a common problem for noise-reference signals and is “particularly problematic” for speech signals. Ex. 1006, 88; Ex. 1003, ¶85. McCowan teaches a technique for improving noise cancellation for near-field speech using **near-field compensation** in the signal path for generating a noise reference signal via a **blocking matrix**, as depicted in Figure 3, below. Ex. 1006, 87, 90-91. The **blocking matrix** “block[s] the desired signal from the adaptive noise estimate,” which requires that the desired signal “be time aligned and have equal amplitudes across all channels.” Ex. 1006, 93.

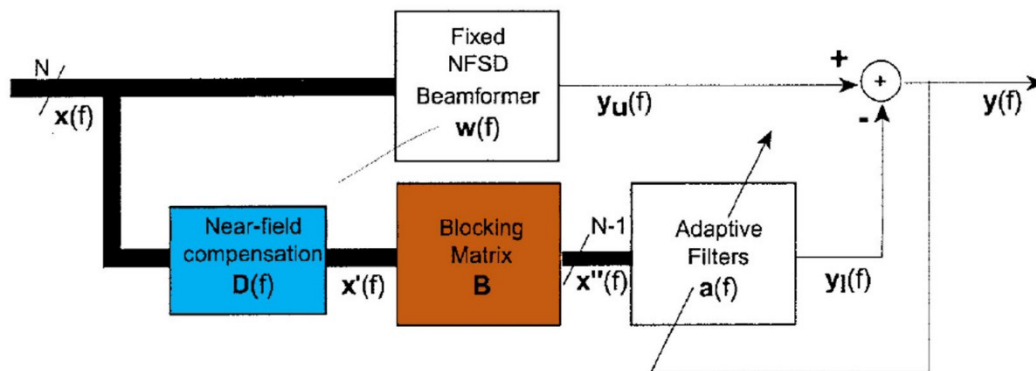


FIG. 3. Near-field adaptive beamformer.

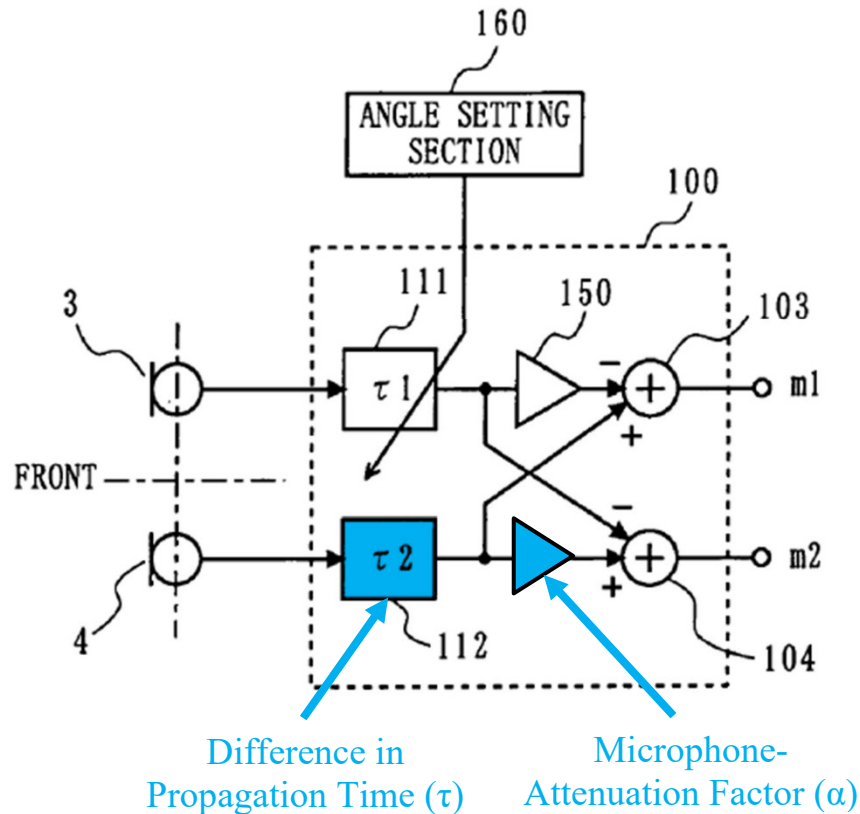
Ex. 1006, Fig. 3 (annotated).

McCowan explains that a spherical wavefront model is “more accurate” for near-field speech than a planar signal wavefront model and “must be employed.” Ex. 1006, 89. “For 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.” Ex. 1006, 93. McCowan discloses two issues that **near-field compensation** must account for to “ensure[] that a true null exists” in the response of the **blocking matrix** “in the direction *and* distance corresponding to the desired source”: (1) “phase misalignment” and (2) “amplitude scaling.” Ex. 1006, 89-90, 93-94; Ex. 1003, ¶86. McCowan discloses that this requires (1) delaying the reference signal and (2) multiplying the reference signal by a calculated gain, where both the length of the delay (the difference in propagation time) and the size of the gain are based on distances between each microphone and the speech source. Ex. 1006, 89-90; Ex. 1003, ¶¶87-89; *supra* §III.B.

Performing **near-field compensation** allows the **blocking matrix** to completely “block the desired signal from the adaptive noise estimate” because the desired signal is both time aligned and has equal amplitudes across the channels. Ex. 1006, 91, 93; Ex. 1003, ¶89. The two channels will be equal in magnitude with respect to “the direction *and* distance corresponding to the desired source” and thus, when subtracting one from the other, the signals cancel each other out, producing “a true null.” Ex. 1006, 93-94; Ex. 1003, ¶89.

It would have been obvious to modify Kanamori's noise reference signal m2 to incorporate McCowan's near-field compensation to address situations when the speech source is in the near field (e.g., a headset). Ex. 1003, ¶90. A POSITA would have been motivated to make such a combination because it would "ensure[] that a true null exists" in m2's directivity pattern "in the direction *and* distance corresponding to the desired source," thereby substantially improving noise suppression in such situations by avoiding the "detrimental effects" caused by speech entering m2. Ex. 1003, ¶90; Ex. 1005, ¶[0083]; Ex. 1006, 93-94. Moreover, such a combination would require minimal modification to Kanamori's Figure 17A, as depicted below. Ex. 1003, ¶90; Ex. 1006, 89-90, 93.

FIG. 17A



Ex. 1005, Fig. 17A (annotated) Ex. 1003, ¶90.

Kanamori already discloses delaying the first microphone signal by d/c , which is the time it takes for sound to travel from one microphone to the other along the axis of the microphones, i.e., the difference in propagation time. Ex. 1005, ¶[0175]; Ex. 1003, ¶91; *supra* §III.C.3 ([1B]). When the speech source originates from a direction of 0° , as discussed above, Kanamori's delay $\tau 2$ and McCowan's difference in propagation time are identical because the distance from microphone unit 3 to the speech source minus the distance from microphone unit 4 to the speech source will equal the distance between microphone units 3 and 4. Ex. 1003, ¶91; Ex. 1006, 89-

90. Accordingly, while a POSITA would be motivated to modify Kanamori's delay τ_2 to McCowan's difference in propagation time to account for near-field speech originating from other directions, Kanamori's delay τ_2 requires no modification for near-field speech at 0° . Ex. 1003, ¶91. In this situation, m_1 is completely unaffected. Ex. 1003, ¶91.

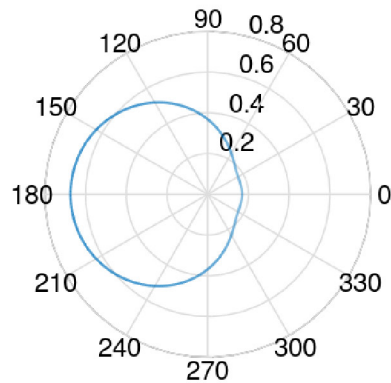
Moreover, Kanamori already discloses signal amplifying section 150 for main signal m_1 . Ex. 1005, ¶¶[0169], [0174]. Kanamori states that its “signal amplification ratio [is] set at approximately 0.85” to create imbalance in the signals subtracted at signal subtracting section 103, thereby increasing the sensitivity in the direction of speech. Ex. 1005, ¶¶[0166], [0169]; Ex. 1003, ¶92. Unlike m_1 , however, a POSITA would have been motivated to create a “true null” in the directivity pattern of m_2 corresponding to “the distance *and* direction” of the speech source to avoid the “detrimental effects” of speech entering the noise reference signal. Ex. 1003, ¶92; Ex. 1005, ¶[0083], Ex. 1006, 93-94. To create this null, McCowan teaches multiplying the delayed signal by a gain (microphone-attenuation factor) to account for the difference in amplitude decay caused by the different distances between the speech source and the microphone units. Ex. 1006, 90, 93-94; Ex. 1003, ¶92. A POSITA would have been motivated to add this microphone-attenuation factor between signal delaying section 112 and signal subtracting section 104 so that “the desired signal” (speech) is both “time aligned and ha[s] equal amplitudes across

[both] channels” entering signal subtracting section 104, ensuring “complete blocking” of the speech signal, as taught by McCowan. Ex. 1006, 93-94; Ex. 1003, ¶92. Adding another signal amplifying section corresponding to McCowan’s microphone-attenuation factor for noise reference signal m2 would therefore have been routine and obvious to account for the amplitude decay due to the different distances traveled. Ex. 1003, ¶92; Ex. 1005, ¶[0169]; Ex. 1006, 90.

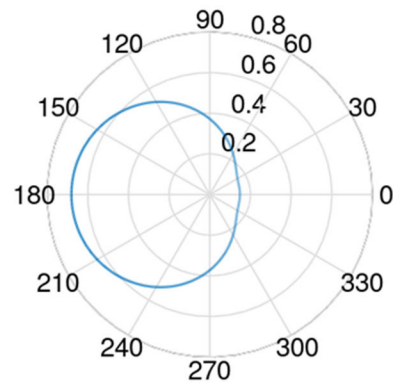
In sum, McCowan teaches incorporating [near-field compensation](#) to align the speech signal in both channels entering signal subtracting section 104 in Kanamori. Ex. 1006, 93; Ex. 1003, ¶93. A POSITA would understand that this would produce a null in m2’s linear response to near-field speech. Ex. 1006, 94; Ex. 1003, ¶93. This aligns with Kanamori’s objective of suppressing “the inclusion of the target sound in the noise reference signal.” Ex. 1005, ¶[0083]; Ex. 1003, ¶93.

Using the modified Figure 17A above incorporating [near-field compensation](#), Dr. Vipperman simulated and plotted m1’s and m2’s linear responses, respectively, to a 1000 Hz signal originating 1 m (noise) from the microphone array, assuming a speech direction of 0°:

m1, Modified Kanamori Fig. 17a at 1m



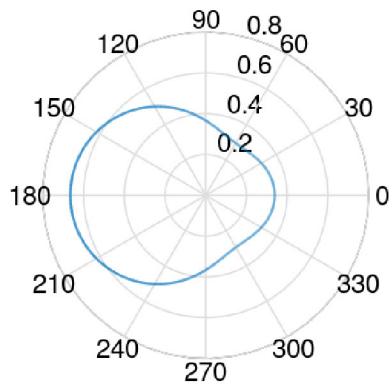
m2, Modified Kanamori Fig. 17a at 1m



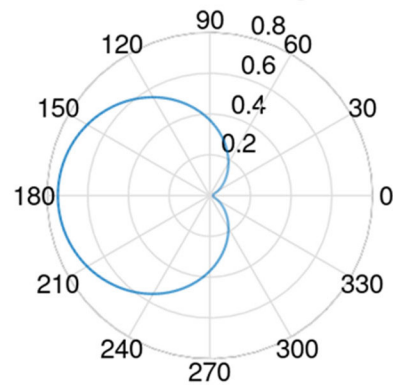
Ex. 1003, ¶¶94. These plots show that m1’s linear response to noise is substantially similar to m2’s linear response to noise. Ex. 1003, ¶¶58, 94. This aligns with Kanamori’s objective that there is no difference in the sensitivity between m1 and m2 for noise “irrespective of the direction from which noise is coming.” Ex. 1005, ¶[0168]; Ex. 1003, ¶94.

Dr. Vipperman also produced the following plots depicting m1’s and m2’s linear responses, respectively, to a 1000 Hz signal originating 0.1 m (speech) from the microphone array, assuming a speech direction of 0°:

m1, Modified Kanamori Fig. 17a at 0.1m



m2, Modified Kanamori Fig. 17a at 0.1m



Ex. 1003, ¶¶95. These plots show that m1’s linear response to speech is substantially dissimilar to m2’s linear response to speech in the direction of speech (0°). Ex. 1003, ¶¶58, 95. This aligns with Kanamori’s objective of including “a large difference in sensitivity between the signals m1 and m2” in the direction of speech. Ex. 1005, ¶[0168]; Ex. 1003, ¶95.

A POSITA would have been motivated to make such a modification because, unlike Kanamori’s Figure 17A, it would allow the cancellation of noise originating from the same direction but at a different distance than the speech. Ex. 1003, ¶96; *see* Ex. 1008 (Dugger), ¶[0035] (disclosing that compensating for the speaker being in the near field “allows preservation of a desired source such as the [speaker] based on distance as well as direction relative to [the microphone array], and provides additional attenuation of the [noise source] in the same direction as the speaker, but at a different distance from the microphone array than the [speaker]”). A POSITA would have been motivated to adjust the linear responses as described above so that

the linear responses to noise of main signal m1 and noise reference signal m2 are substantially similar in all directions, making the estimated noise spectrum accurate in all directions, including the direction of the target speech, while maintaining Kanamori's teaching that the linear responses to speech of m1 and m2 should be different in the direction of the speech source. Ex. 1003, ¶96; Ex. 1005, ¶¶[0168], [0103]-[0106] (explaining how the estimated noise spectrum affects noise cancellation).

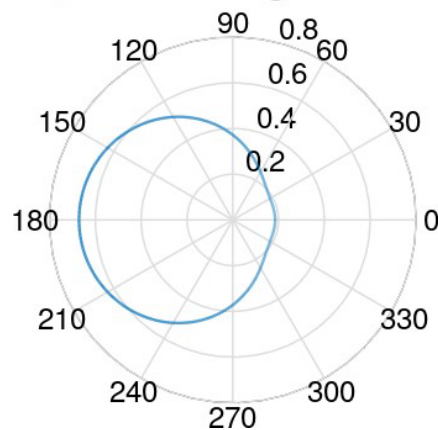
A POSITA would have had a reasonable expectation of success in making such a combination because it is a combination of familiar elements according to known methods to yield predictable results. Ex. 1003, ¶97. Kanamori already discloses techniques for varying the delays, including τ_2 , providing a reasonable expectation that similar techniques could be used to adjust the difference in propagation time to account for different speaker positions. Ex. 1005, ¶¶[0175], [0180], Figs. 17A, 18A; Ex. 1003, ¶97. Likewise, Kanamori already discloses signal amplifying section 150, providing a reasonable expectation that a similar amplifying section could be added to account for amplitude decay. Ex. 1005, ¶[0169]; Ex. 1003, ¶97. Moreover, similar to Kanamori's noise reference signal m2, McCowan discloses using this technique to generate a "noise estimate" that completely blocks the desired signal. Ex. 1006, 93; Ex. 1003, ¶97.

9. [1H]: “wherein at least one of the first linear response to noise and the second linear response to noise is being non-zero in a direction towards a source of noise.”

Kanamori discloses or suggests [1H].

As explained in Section III.A.3.c ([1B]), Kanamori discloses the following first linear response to noise:

m1, Kanamori Fig. 17a at 1m



Ex. 1003, ¶98. As depicted, m1’s first linear response to noise is non-zero in all directions and therefore is non-zero in a direction towards a source of noise. Ex. 1003, ¶98; *see also* Ex. 1005, ¶[0167] (disclosing that m1 includes “components of the sound coming from a direction other than the direction of the target sound more than components of the target sound”).

D. Dependent Claims 2-22

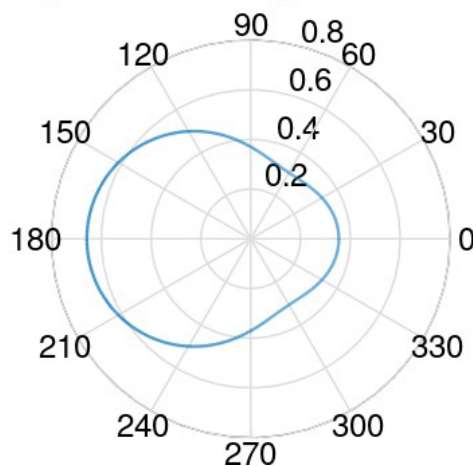
1. **Claim 2: “The microphone array of claim 1, wherein the first and second physical microphones are omnidirectional.”**

Kanamori discloses or renders obvious that the first and second physical microphones are omnidirectional. *Supra* §§III.C.2 ([1A]), III.C.5 ([1D]); Ex. 1003, ¶99.

2. **Claim 3: “The microphone array of claim 1, wherein the first linear response to speech is devoid of a null, wherein the speech is human speech.”**

Kanamori discloses or suggests this additional limitation. As explained *supra* §III.C.3 ([1B]), Kanamori discloses that m1 has the following first linear response to near-field speech:

m1, Kanamori Fig. 17a at 0.1m



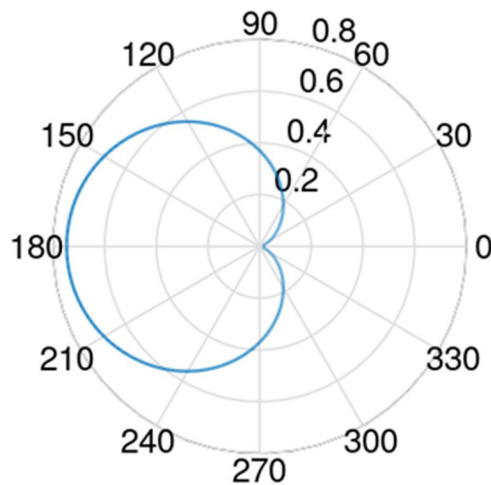
Ex. 1003, ¶100. As depicted, m1’s linear response to near-field speech lacks a null. Ex. 1003, ¶100.

Additionally, Kanamori discloses that the speech is human speech. Kanamori discloses using its microphone device to receive “voice from the talker.” Ex. 1005, ¶[0013]; *see also* Ex. 1005, ¶[0125], Fig. 9 (disclosing a “talker” approximately 50 cm from a microphone array). A POSITA would understand that “voice from the talker” refers to human speech. Ex. 1005, ¶[0013]; Ex. 1003, ¶101.

3. Claims 4-7

Kanamori and McCowan render obvious the additional limitations of claims 4-7. As explained *supra* Section III.C.8 ([1G]), it would be obvious to modify Kanamori’s noise reference signal m2 (the second virtual microphone) to produce the following second linear response to speech:

m2, Modified Kanamori Fig. 17a at 0.1m



Ex. 1003, ¶102.

- a. **Claim 4: “The microphone array of claim 3, wherein the second linear response to speech of the second virtual microphone includes a single null oriented in a direction toward a source of the speech.”**

As depicted, m2’s linear response to speech includes a single null oriented at 0°, which in the above plot is the direction toward a source of the speech. Ex. 1003, ¶¶58, 103; *supra* §III.C.8 ([1G]).

- b. **Claim 5: “The microphone array of claim 4, wherein the single null is a region of the second linear response to speech having a measured response level that is lower than the measured response level of any other region of the second linear response to speech.”**

As depicted, the null at 0° represents the lowest measured response level of any region in m2’s linear response to speech. Ex. 1003, ¶¶58, 104; *supra* §III.C.8 ([1G]); *see* Ex. 1005, ¶[0167] (disclosing that m2’s direction of minimum sensitivity points towards speech).

- c. **Claim 6: “The microphone array of claim 4, wherein the second linear response to speech includes a primary lobe oriented in a direction away from the source of the speech.”**

As depicted, the primary lobe of m2’s linear response is oriented towards 180°, away from the source of speech at 0°. Ex. 1003, ¶¶58, 105; *supra* §III.C.8 ([1G]).

- d. Claim 7: “The microphone array of claim 6, wherein the primary lobe is a region of the second linear response to speech having a measured response level that is greater than the measured response level of any other region of the second linear response to speech.”**

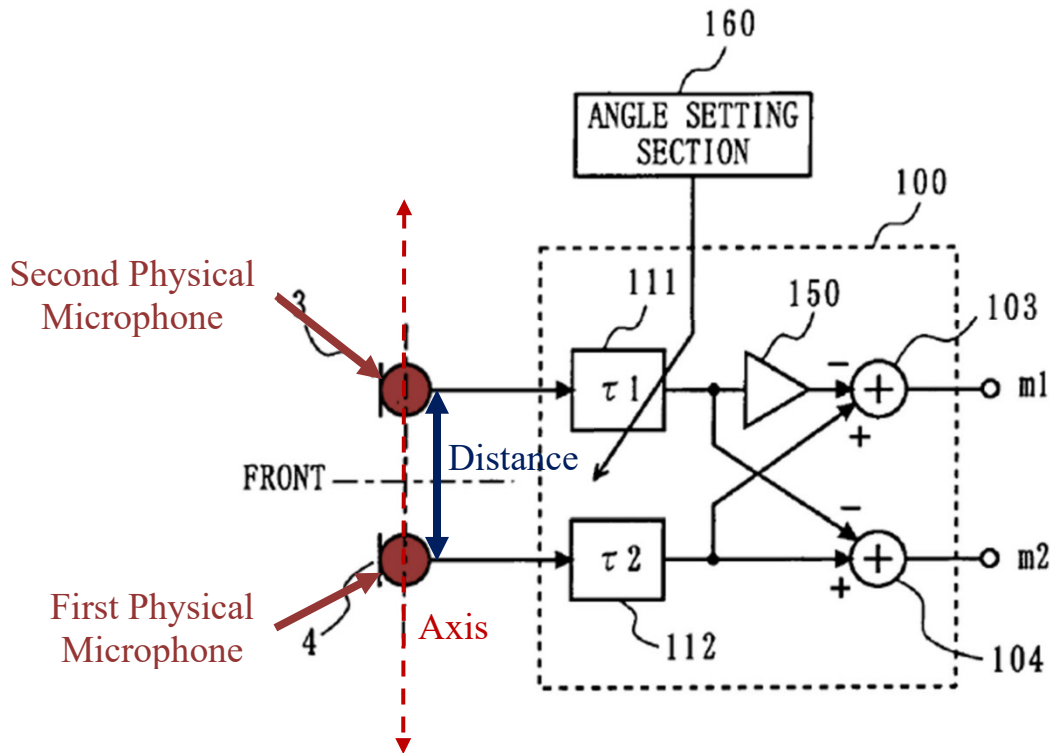
As depicted, the primary lobe of m2’s linear response to speech is oriented towards 180° and is the region having the highest measured response level of any region of the second linear response to speech. Ex. 1003, ¶¶58, 106; *supra* §III.C.8 ([1G]).

- 4. Claim 8: “The microphone array of claim 4, wherein the first physical microphone and the second physical microphone are positioned along an axis and separated by a first distance.”**

Kanamori discloses or suggests this additional limitation. Ex. 1003, ¶107.

Kanamori discloses that the “microphone units 3 and 4 are placed on an axis (denoted by a one-dot chain line . . .) perpendicular to a straight line oriented to the front.” Ex. 1005, ¶[0160]. Kanamori also discloses that “the microphone units are required to be a certain distance (normally, 1 cm to 5 cm) apart from each other.” Ex. 1005, ¶[0151].

FIG. 17A



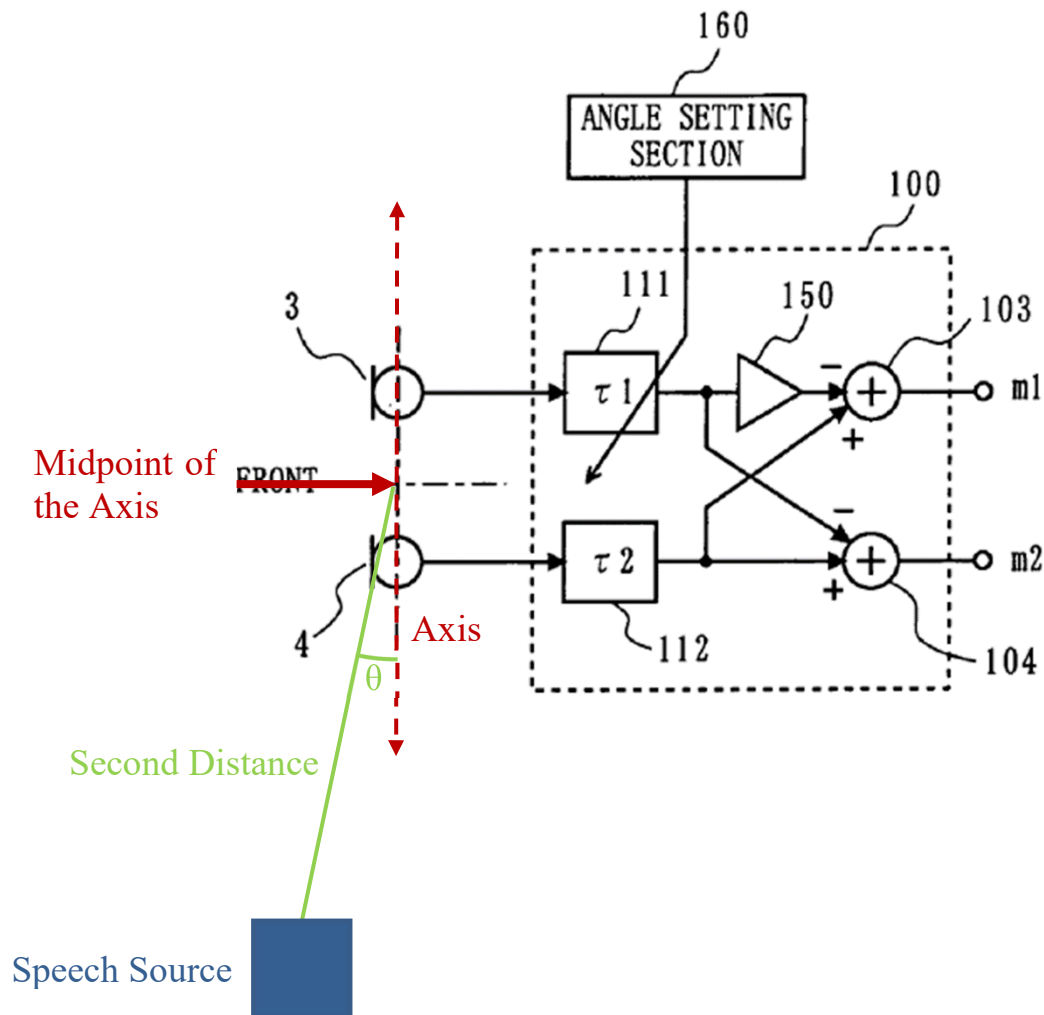
Ex. 1005, Fig. 17A (annotated).

5. **Claim 9:** “The microphone array of claim 8, wherein a midpoint of the axis is a second distance from the speech source that generates the speech, wherein the speech source is located in a direction defined by an angle relative to the midpoint.”

Kanamori discloses or suggests this additional limitation. Kanamori discloses that the microphone units are “placed on an **axis**” and “required to be a certain distance (normally, 1 cm to 5 cm) apart from each other.” Ex. 1005, ¶¶[0151], [0160]; *supra* §III.D.4 (claim 8). A midpoint of the **axis** is located halfway between the microphones along that **axis**. Ex. 1003, ¶108. This midpoint is necessarily located

at a distance (**second distance**) from the speech source. Ex. 1003, ¶108. Kanamori discloses an “angle setting section” that “can change a signal delay amount τ_1 ” so that “the target sound direction can be changed in a range of ± 90 degrees with respect to the front direction.” Ex. 1005, ¶[0175]. A POSITA would understand that the speech source is located in a direction defined by an angle relative to the midpoint of the **axis**. Ex. 1003, ¶108.

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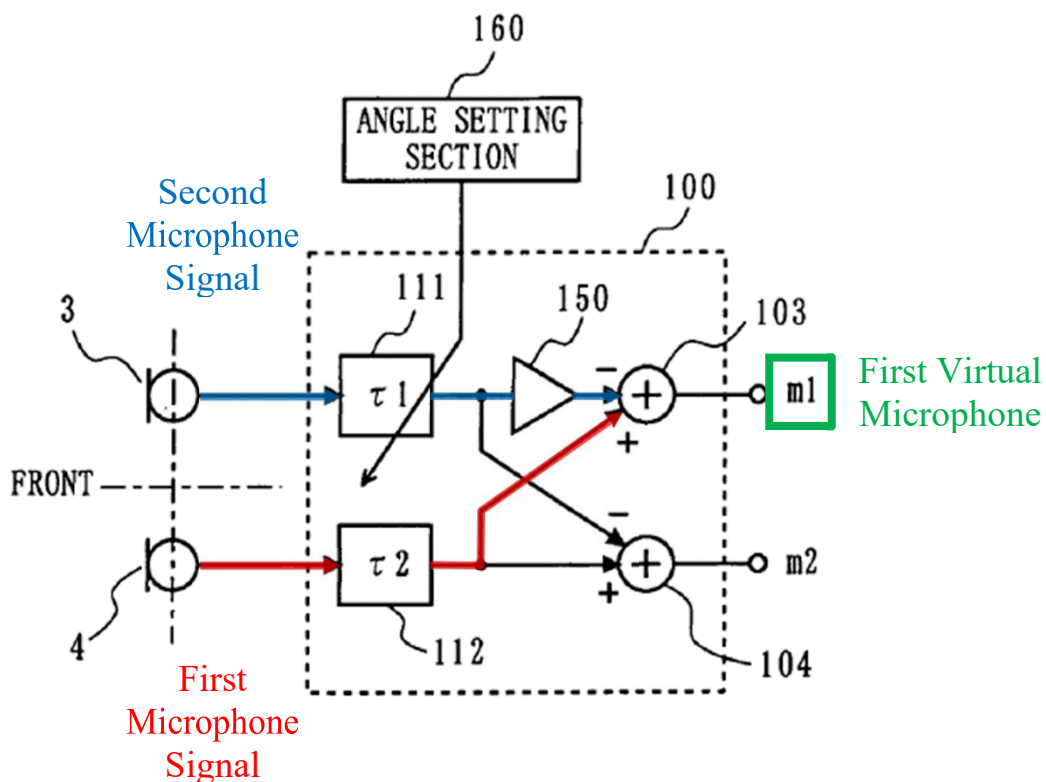


Ex. 1005, Fig. 17A (annotated). In Dr. Vipperman's simulations, the speech source is located at a second distance of 0.1 m and an angle of 0°. *Supra* §§III.C.3-III.C.4 ([1B]-[1C]); Ex. 1003, ¶108.

6. **Claim 10: “The microphone array of claim 9, wherein the first virtual microphone comprises the second microphone signal subtracted from the first microphone signal.”**

Kanamori discloses or suggests this additional limitation. As explained for [1A], Kanamori's signal subtracting section 103 subtracts the **second microphone signal** from the **first microphone signal** to obtain **first virtual microphone m1**. *Supra* §III.C.2; Ex. 1003, ¶109.

F I G . 1 7 A



Ex. 1005, Fig. 17A (annotated).

**7. Claim 11: “The microphone array of claim 10,
wherein the first microphone signal is delayed.”**

Kanamori discloses or Kanamori and McCowan render obvious this additional limitation. Ex. 1003, ¶110. Kanamori discloses that a “**signal output from the fourth microphone unit**” (the first microphone signal) is delayed by “the second signal delaying section 112.” Ex. 1005, ¶¶[0166], [0174]; *supra* §III.C.2 ([1A]). Additionally, it would have been obvious to modify the delay to account for phase misalignment in view of McCowan. *Supra* §III.C.8 ([1G]).

**8. Claim 12: “The microphone array of claim 11,
wherein the delay is raised to a power that is
proportional to a time difference between arrival of
the speech at the first virtual microphone and arrival
of the speech at the second virtual microphone.”**

The '691 patent defines a value γ as follows:

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

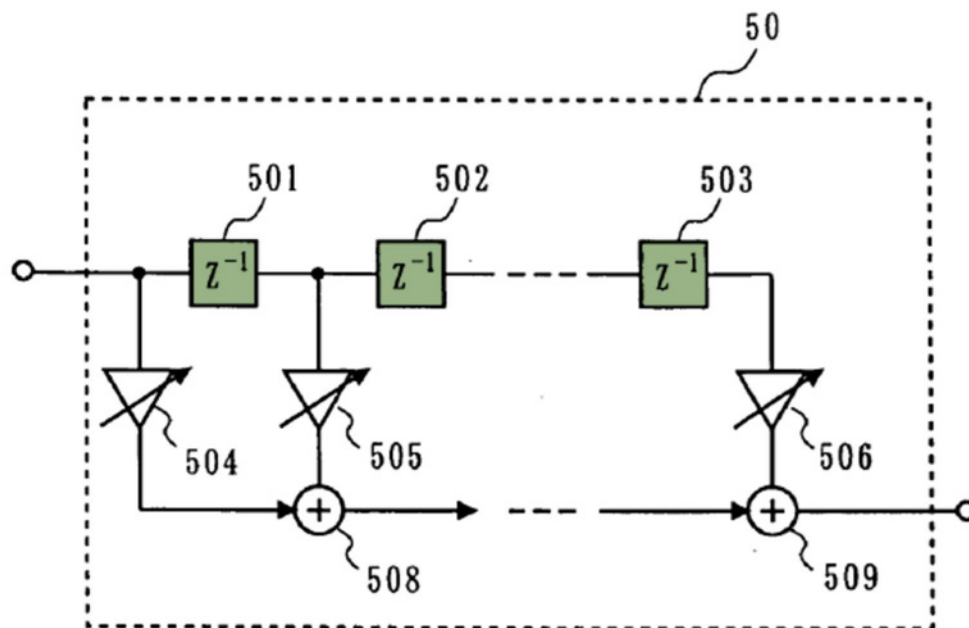
where “[t]he distances d_1 and d_2 are the distance from O_1 and O_2 to the speech source . . . , and γ is their difference divided by c , the speed of sound, and multiplied by the sampling frequency f_s .” Ex. 1001, 10:50-11:4. The '691 patent later refers to γ as “the time difference between arrival of speech at V_1 compared to V_2 .” Ex. 1001, 14:66-15:2.

To the extent this claim is not indefinite, and based on the '691 patent's reference to γ as “the time difference between arrival of speech at V_1 compared to V_2 ,” Ex. 1001, 14:66-15:2, which Petitioner adopts for the purpose of this IPR only, Kanamori discloses or Kanamori and McCowan render obvious this additional limitation. Kanamori discloses that delay τ_2 at signal delaying section 112 is d/c , “where d is a distance between the microphone units and c is the speed of sound.” Ex. 1005, ¶[0175]. As explained for [1G], when the speech originates from a direction of 0° , d/c is the time difference between arrival of the speech at microphone unit 3 and microphone unit 4. Ex. 1003, ¶112; *supra* §III.C.8 ([1G]). Additionally, as explained for [1G], it would have been obvious to modify delay τ_2 to McCowan's difference in propagation time, which, with reference to Figure 17A, is the distance from microphone unit 3 to the speech source minus the distance from microphone unit 4 to the speech source divided by the speed of sound. Ex. 1003, ¶112; Ex. 1006, 89-90; *supra* §III.C.8 ([1G]).

Kanamori discloses that “a digital process can be used” for the “directivity combining process.” Ex. 1005, ¶[0151]. In digital processes, signals are represented by discrete samples, where the number of samples over a particular time period of the signal depends on the sampling frequency (signal samples per unit time). Ex. 1003, ¶113; *see* Ex. 1005, ¶[0108]. For digital processes in the microphone-array

art, delays are commonly expressed as z^{-s} , where s constitutes the numbers of samples by which the process is delayed. Ex. 1003, ¶¶60, 113. For example, Figure 5 of Kanamori discloses several signal delaying sections 501, 502, and 503, where “each signal delaying section is connected in series to each other for delaying a received signal by one sample.” Ex. 1005, ¶[0108], Fig. 5.

F I G. 5



Ex. 1005, Fig. 5 (annotated). In Figure 5, z^{-1} refers to a delay by one sample. Ex. 1005, ¶[0108], Fig. 5. A POSITA would understand that determining how many samples exist over a particular time period is a simple matter of multiplying the time period by the sampling frequency. Ex. 1003, ¶113. Because the signal combining

process is a digital one, a POSITA would therefore understand that delaying the signal output from microphone unit 4 by time τ_2 or the difference in propagation time refers to delaying this signal by the number of samples corresponding to the delayed time period, which is expressed as

$$z^{-\tau \cdot f_s},$$

where τ is the desired time delay and f_s is the sampling frequency. Ex. 1003, ¶113.

Accordingly, a POSITA would understand that, by delaying the first microphone signal by the time it takes for a soundwave to travel the distance between the microphones (either via Kanamori's τ_2 or McCowan's difference in propagation time), the delay z is raised to a power that is proportional to a time difference between arrival of the speech at the first virtual microphone and arrival of the speech at the second virtual microphone. Ex. 1003, ¶114.

- 9. Claim 13: “The microphone array of claim 11, wherein the delay is raised to a power that is proportional to a sampling frequency multiplied by a quantity equal to a third distance subtracted from a fourth distance, the third distance being between the first physical microphone and the speech source and the fourth distance being between the second physical microphone and the speech source.”**

Kanamori discloses or Kanamori and McCowan render obvious this additional limitation. As explained for claim 12, Kanamori discloses, or Kanamori and McCowan render obvious, delaying the signal output from microphone unit 4 by:

$$z^{-\tau} f_s,$$

where τ is the desired time delay (either Kanamori's τ_2 or McCowan's difference in propagation time) and f_s is the sampling frequency. Ex. 1003, ¶115; *supra* §III.D.8. As explained for [1G], when the speech source is at 0° , τ_2 is proportional to a distance between the first physical microphone and the speech source subtracted from a distance between the second physical microphone and the speech source. Ex. 1003, ¶115; *supra* §III.C.8. Also as explained for [1G], McCowan's difference in propagation time is proportional to a distance between the first physical microphone and the speech source subtracted from a distance between the second physical microphone and the speech source. Ex. 1003, ¶115; *supra* §III.C.8.

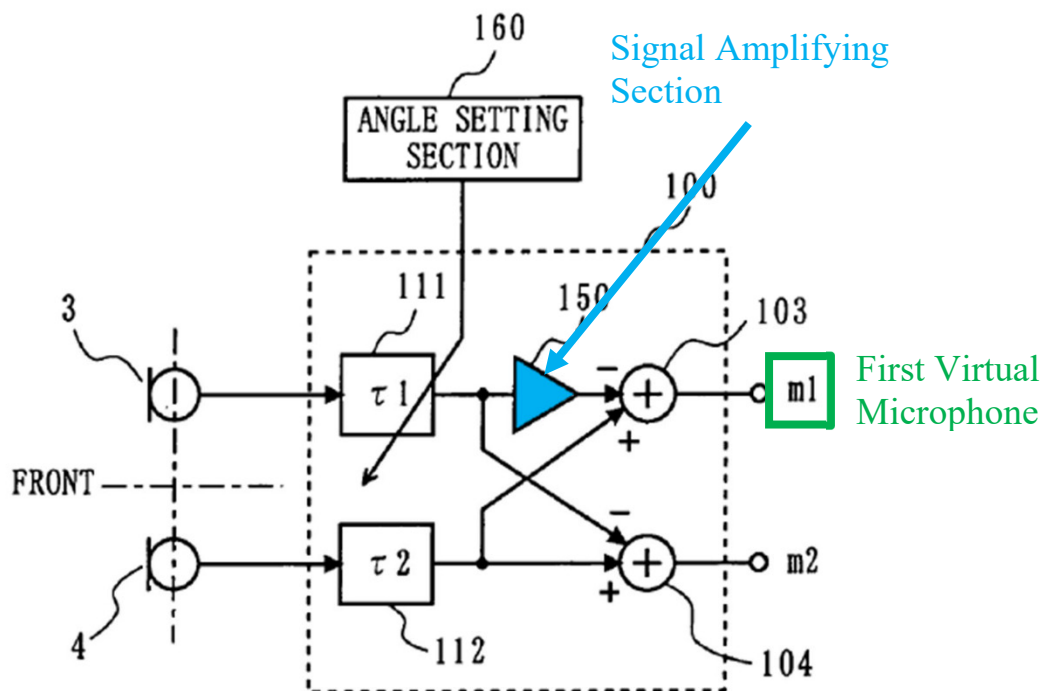
Accordingly, a POSITA would understand that Kanamori discloses, or Kanamori and McCowan render obvious, a delay raised to a power that is proportional to a sampling frequency f_s multiplied by a quantity equal to a third distance subtracted from a fourth distance, the third distance being between the first physical microphone and the speech source and the fourth distance being between the second physical microphone and the speech source. Ex. 1003, ¶116.

10. **Claim 14:** “The microphone array of claim 10, wherein the second microphone signal is multiplied by a ratio, wherein the ratio is a ratio of a third distance to a fourth distance, the third distance being between the first physical microphone and the speech source and the fourth distance being between the second physical microphone and the speech source.”

Kanamori and McCowan render obvious this additional limitation.

Kanamori discloses a [signal amplifying section 150](#) for [main signal m1](#). Ex. 1005, ¶¶[0169], [0174]. Kanamori states that its “signal amplification ratio [is] set at approximately 0.85” to create imbalance in the signals that are subtracted at signal subtracting section 103, thereby increasing the sensitivity in the direction of speech. Ex. 1005, ¶¶[0166], [0169]; Ex. 1003, ¶117.

FIG. 17A



Ex. 1005, Fig. 17A (annotated).

Kanamori includes this [signal amplifying section](#) because, assuming a far-field speech source where the differences in amplitude decay are negligible across the microphones, phase-aligned speech signals in the two channels could cancel each other out at signal subtracting section 103, producing a null in the directivity pattern of [m1](#) in the direction of the speech source. Ex. 1003, ¶118; Ex. 1005, Fig. 17B (depicting null in [m2](#)'s directivity pattern in the direction of the speech source); Ex. 1006, 90; *supra* §III.C.8 ([1G]). This would dramatically reduce the efficacy of Kanamori's noise-suppression technique, which is premised on "suppress[ing], based on the noise reference signal, noise components included in the main signal under the circumstances where noise is coming from a plurality of directions." Ex. 1005, ¶[0168]; Ex. 1003, ¶118. If [m1](#) did not include the speech components due to a null in the direction of speech, Kanamori's system would not work. Ex. 1003, ¶118.

It would have been obvious to combine Kanamori and McCowan to implement McCowan's near-field compensation for when the speech source is in the near field. *Supra* §III.C.8 ([1G]). As further discussed for [1G], it would have been obvious to modify Kanamori's τ_2 to implement McCowan's τ_i to account for near-

field speech originating from other directions. *Id.* This aligns with Kanamori’s disclosure that Figure 17A can change the target sound direction “in a range of ± 90 degrees with respect to the front direction.” Ex. 1005, ¶[0175]; Ex. 1003, ¶119.

Unlike far-field speech, the “resulting amplitude differences across sensors” from wavefront amplitude decay “can be significant in the near-field case.” Ex. 1006, 90. Because of amplitude decay, a POSITA would have been motivated to modify Kanamori’s signal amplifying section 150 to implement McCowan’s microphone-attenuation factor α_i to ensure that speech is not attenuated or completely canceled in **main signal m1** when approaching from different directions. Ex. 1003, ¶120; Ex. 1006, 90. For example, as discussed for [1G], speech may approach the microphone array from 0° . *Supra* §III.C.8 ([1G]). In such a situation, the speech would travel a shorter distance to microphone unit 4 than it would to microphone unit 3, meaning the speech signal would have a greater amplitude at microphone unit 4 than it would at microphone unit 3. Ex. 1003, ¶120; Ex. 1006, 90. In such a situation, Kanamori’s amplification ratio of 0.85 would create further imbalance in the two channels. Ex. 1003, ¶120.

If, however, speech approached the microphone array from 180° , the speech would travel a shorter distance to microphone unit 3 than it would to microphone unit 4, meaning the speech signal would have a greater amplitude at microphone unit

3 than it would at microphone unit 4. Ex. 1003, ¶121; Ex. 1006, 90. Here, Kana-mori’s amplification ratio would not necessarily ensure that the two channels are imbalanced as they enter signal subtracting section 103 because the larger speech amplitude in the upper channel of Figure 17A would be reduced by a factor of 0.85, driving its amplitude toward the smaller speech amplitude in the lower channel of Figure 17A. Ex. 1003, ¶¶121-122; Ex. 1005, ¶[0169], Fig. 17A.

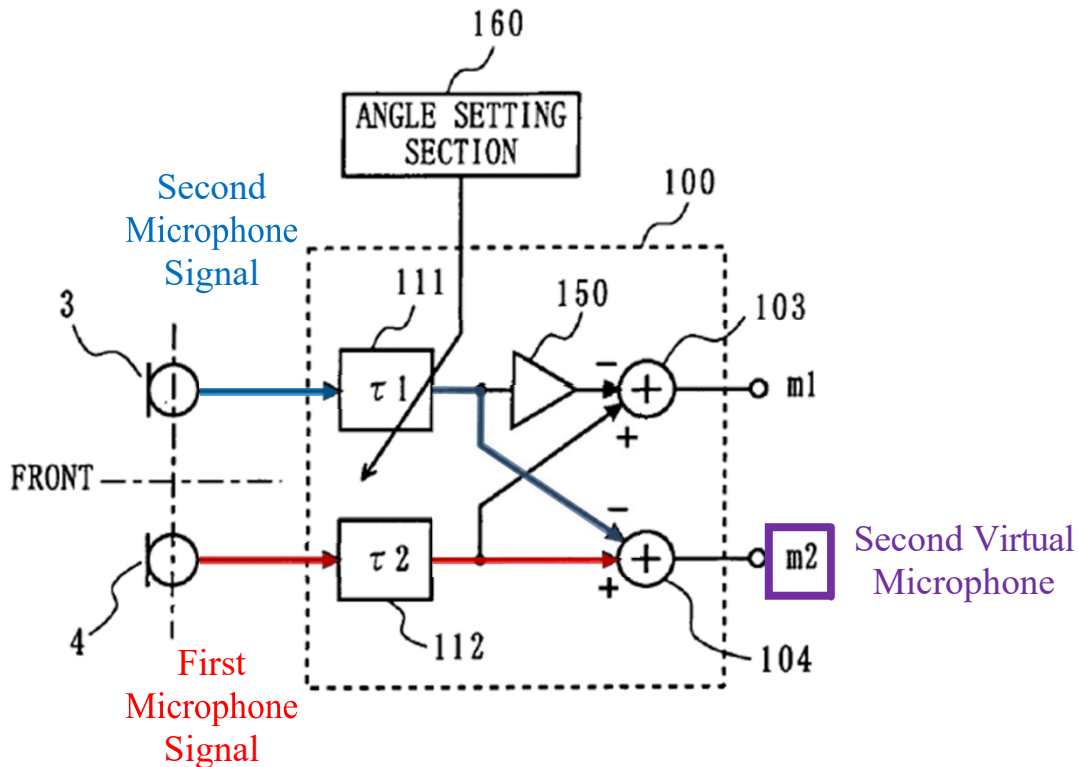
A POSITA would have been motivated to modify signal amplifying section 150 in Figure 17A to multiply “the signal output from the signal delaying section 111” by McCowan’s microphone-attenuation factor α_i , should the microphone device instead anticipate speech from a direction other than 0°. Ex. 1003, ¶123; Ex. 1005, ¶[0166]. If the speech source approached the array from an angle greater than 90°, it would be closer to microphone unit 3, causing the microphone-attenuation factor to be greater than 1. Ex. 1003, ¶123; Ex. 1006, 90. If it approached the array from an angle less than 90°, it would be closer to microphone unit 4, causing the microphone-attenuation factor to be less than 1. Ex. 1003, ¶123; Ex. 1006, 90. Modifying the amplification ratio of [signal amplifying section 150](#) to McCowan’s microphone-attenuation factor would therefore ensure that the microphone device could be configured to receive speech from any angle between 0° and 180° without risking a null in the direction of speech, improving [m1](#)’s sensitivity to speech. Ex. 1003, ¶123; Ex. 1005, ¶¶[0169], [0175]; Ex. 1006, 90.

A POSITA would have had a reasonable expectation of success in making such a combination because it is a combination of familiar elements according to known methods to yield predictable results. Ex. 1003, ¶124. Kanamori already discloses techniques for varying aspects of the directivity combining section 100, such as the value of delay τ_1 , to account for a target sound originating from various directions, providing a reasonable expectation of success of also adjusting the amplification ratio at signal amplifying section 150 based on known and/or expected distances between the speech source and the microphone distances. Ex. 1005, ¶[0175]; Ex. 1003, ¶124. Moreover, a POSITA would have had a reasonable expectation of successfully implementing McCowan’s microphone-attenuation factor for the noise reference signal. *Supra* §III.C.8 ([1G]). It would have been routine for a POSITA to apply this same microphone-attenuation factor for the main signal. Ex. 1003, ¶124.

11. Claim 15: “The microphone array of claim 9, wherein the second virtual microphone comprises the first microphone signal subtracted from the second microphone signal.”

Kanamori renders obvious this additional limitation. Kanamori discloses that the **second virtual microphone** comprises the **second microphone signal** subtracted from the **first microphone signal**. *Supra* §III.C.6 ([1E]).

FIG. 17A



Ex. 1005, Fig. 17A (annotated).

A POSITA would have known that the sign of the output of signal subtracting section 104 (either positive or negative) has no impact on m2's linear responses to sounds at various distances. Ex. 1003, ¶126. Accordingly, switching the positive and negative inputs to signal subtracting section 104 would not affect either the sensitivity or the directivity of m2. Ex. 1003, ¶126. Switching the positive and negative inputs to signal subtracting section 104 would have been obvious to try because there are only two options for which channel enters the positive input and which channel enters the negative input. Ex. 1003, ¶126. Because both arrangements produce

equivalent linear responses and switching the positive and negative inputs requires a simple modification to Figure 17A's structure, it would have been obvious to construct m2 by subtracting the first microphone signal from the second microphone signal. Ex. 1003, ¶126. Moreover, Kanamori discloses various embodiments where the terminals change, further motivating a POSITA to try the alternative arrangement and providing a reasonable expectation of success in achieving this alternative arrangement. *See* Ex. 1005, Figs. 17A, 18A (switching the positive and negative inputs to signal subtracting section 103); Ex. 1003, ¶126.

12. Claim 16: “The microphone array of claim 15, wherein the first microphone signal is delayed.”

Kanamori discloses, or Kanamori and McCowan render obvious, this additional limitation. *Supra* §III.D.7 (claim 11); Ex. 1003, ¶127.

13. Claim 17: “The microphone array of claim 16, wherein the delay is raised to a power that is proportional to a time difference between arrival of the speech at the first virtual microphone and arrival of the speech at the second virtual microphone.”

Kanamori discloses, or Kanamori and McCowan render obvious, this additional limitation. *Supra* §III.D.8(claim 12); Ex. 1003, ¶128.

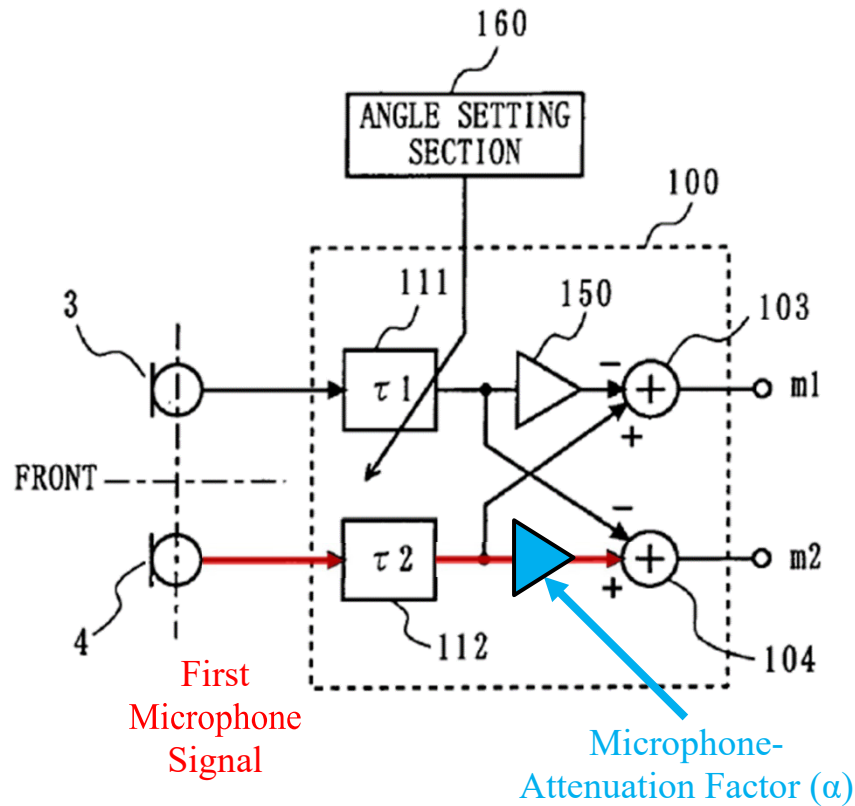
14. **Claim 18: “The microphone array of claim 16, wherein the power is proportional to a sampling frequency multiplied by a quantity equal to a third distance subtracted from a fourth distance, the third distance being between the first physical microphone and the speech source and the fourth distance being between the second physical microphone and the speech source.”**

Kanamori discloses, or Kanamori and McCowan render obvious, this additional limitation. *Supra* §III.D.9 (claim 13); Ex. 1003, ¶129.

15. **Claim 19: “The microphone array of claim 18, wherein the first microphone signal is multiplied by a ratio, wherein the ratio is a ratio of the third distance to the fourth distance.”**

Kanamori and McCowan render obvious this additional limitation. Ex. 1003, ¶130. As explained for [1G], it would have been obvious to modify Kanamori to implement McCowan’s **microphone-attenuation factor**, which corresponds to the distance from a reference microphone to the speech signal divided by the distance from the *i*th microphone to the speech signal. *Supra* §III.C.8; Ex. 1006, 90. Implementing this **microphone-attenuation factor** into Kanamori’s Figure 17A, it would have been obvious to multiply the **first microphone signal** by the **microphone-attenuation factor**, which with reference to Figure 17A is the distance from microphone unit 4 to the speech source (third distance) divided by the distance from microphone unit 3 to the speech source (fourth distance). *Supra* §III.C.8 ([1G]).

FIG. 17A

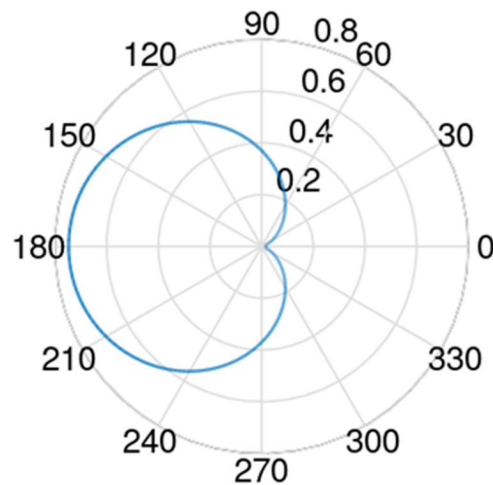


Ex. 1005, Fig. 17A (annotated).

- 16. Claim 20: “The microphone array of claim 4, wherein the single null is located at a distance from at least one of the first physical microphone and the second physical microphone where the source of the speech is expected to be.”**

Kanamori and McCowan render obvious this additional limitation. As explained for [1G] and claim 4, it would have been obvious to modify Kanamori's noise reference signal m2 to include a single null oriented at 0°, which is the direction toward a source of the speech in the plot depicted below. Ex. 1003, ¶131; *supra* §§III.C.8 ([1G]), III.D.3.a (claim 4).

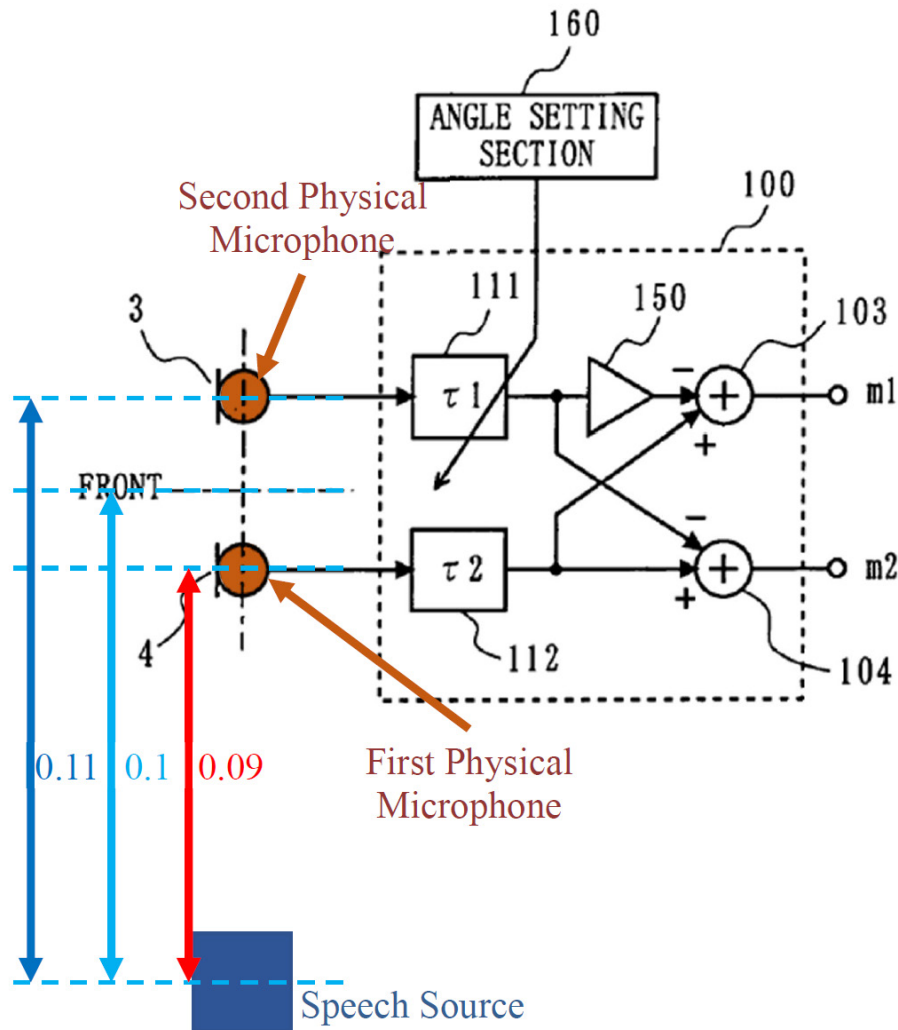
m2, Modified Kanamori Fig. 17a at 0.1m



Ex. 1003, ¶131. In the above plot, the single null is located at a distance of 0.1 m from where the source of the speech is expected to be and the midpoint of the array.

Ex. 1003, ¶131; *supra* §III.C.8 ([1G]). A POSITA would understand that this single null is also located a distance from microphone unit 4 (first physical microphone) and microphone unit 3 (second physical microphone). Ex. 1003, ¶131. For example, if the microphones are spaced 2 cm apart, Ex. 1005, ¶[0151], for a speech source expected at 0° and 0.1 m from the midpoint of the array, the null is located 0.09 m from microphone unit 4 and 0.11 m from microphone unit 3. Ex. 1003, ¶131.

FIG. 17A



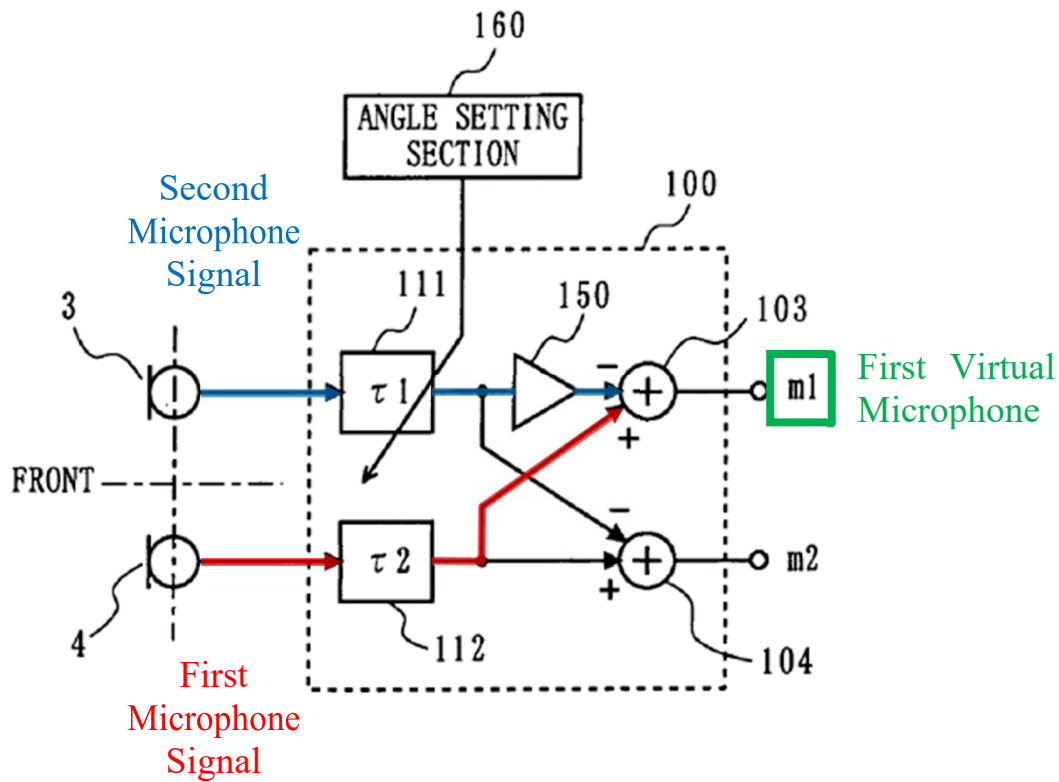
Ex. 1005, Fig. 17A (annotated); Ex. 1003, ¶131.

17. **Claim 21:** “The microphone array of claim 1, wherein the first virtual microphone comprises the second microphone signal subtracted from a delayed version of the first microphone signal.”

Kanamori discloses or suggests this additional limitation. As explained for [1A], main signal m1 constitutes a first virtual microphone that comprises the “signal output from the third microphone unit 3” (second microphone signal) subtracted

from a delayed version of the “signal output from the fourth microphone unit 4” (first microphone signal). Ex. 1005, ¶¶ [0166], [0174]; Ex. 1003, ¶ 132; *supra* §III.C.2.

F I G. 1 7 A



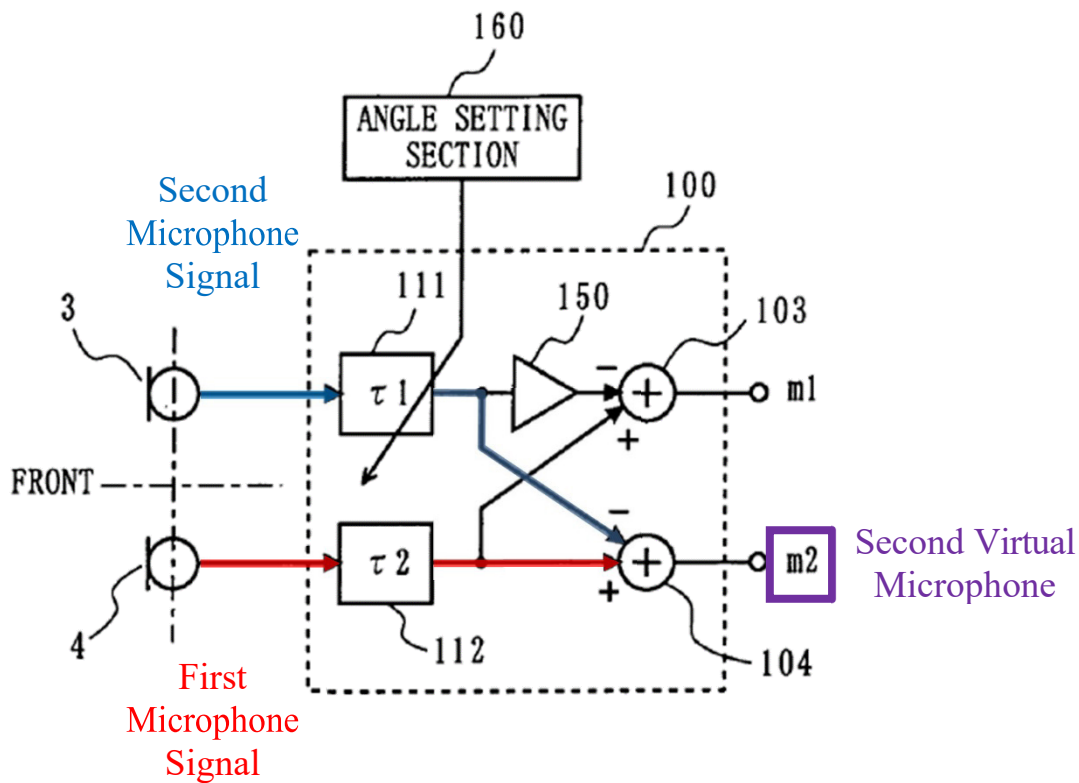
Ex. 1005, Fig. 17A (annotated).

18. **Claim 22:** “The microphone array of claim 21, wherein the second virtual microphone comprises a delayed version of the first microphone signal subtracted from the second microphone signal.”

Kanamori renders obvious this additional limitation. As explained for [1E], noise reference signal m2 constitutes a second virtual microphone that comprises the

“signal output from the third microphone unit 3” (second microphone signal) subtracted from a delayed version of the “signal output from the fourth microphone unit 4” (first microphone signal). Ex. 1005, ¶¶[0166], [0174]; Ex. 1003, ¶133; *supra* §III.C.6.

F I G. 1 7 A



Ex. 1005, Fig. 17A (annotated).

As explained for claim 15, it would have been obvious to switch the positive and negative inputs to signal subtracting section 104 such that the second virtual microphone comprises a delayed version of the first microphone signal subtracted from the second microphone signal. *Supra* §III.D.11 (claim 15); Ex. 1003, ¶134.

E. Independent Claim 29

1. [29P]: “A device comprising:”

Kanamori discloses or suggests [29P].

Kanamori discloses a “microphone device” (e.g., an audio recorder or video recorder) that includes two microphone units spaced a certain distance apart.

Ex. 1005, ¶¶[0077], [0151], [0201], Figs. 14-21, 23. A microphone device is a device. Ex. 1003, ¶135.

2. [29A]: “a first physical microphone generating a first microphone signal;”

Kanamori discloses or suggests [29A] for the reasons explained for [1D]. *Supra* §III.C.5; Ex. 1003, ¶136.

3. [29B]: “a second physical microphone generating a second microphone signal; and”

Kanamori discloses or suggests [29B] for the reasons explained for [1D]. *Supra* §III.C.5; Ex. 1003, ¶137.

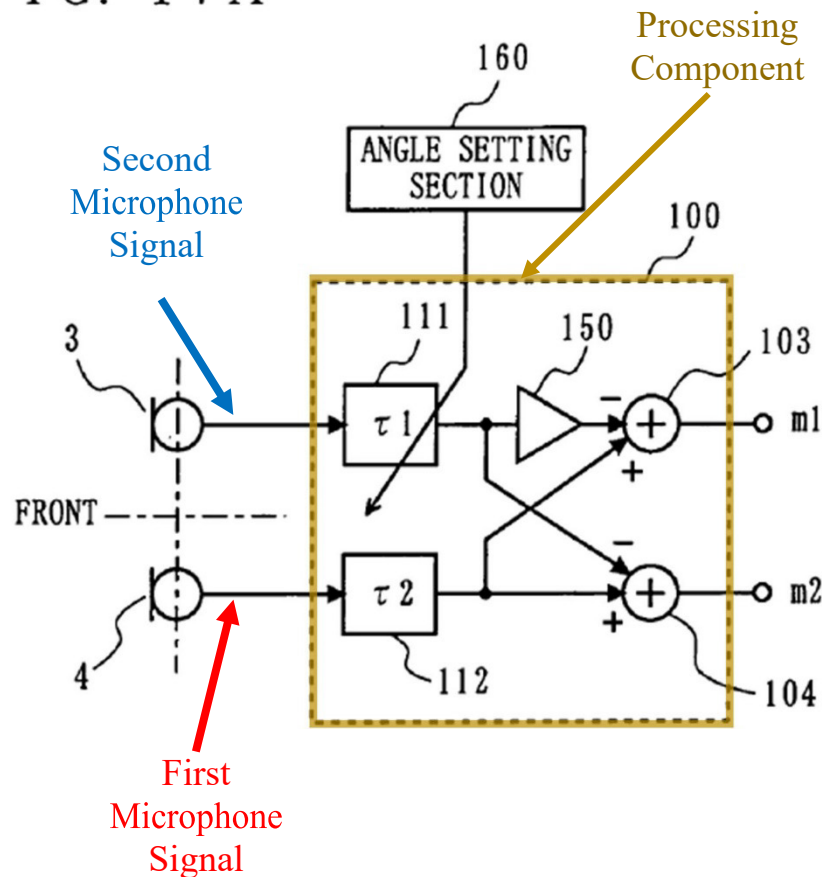
4. [29C]: “a processing component coupled to the first microphone signal and the second microphone signal,”

Kanamori discloses or renders obvious [29C].

Kanamori discloses that a “**directivity combining section 100**” is coupled to the **first microphone signal** and **second microphone signal** and is employed to obtain a main signal m1 and a noise reference signal m2. Ex. 1005, ¶¶[0152], [0173]. A

POSITA would have understood that the **directivity combining section** is a processing component because Kanamori discloses that the “directivity combining process” allows for high flexibility in designing directivity and a variable characteristic using a digital process. Ex. 1005, ¶[0151]; Ex. 1003, ¶138. A POSITA would have understood that implementing a variable characteristic using a digital process requires a processing component to carry out the digital process. Ex. 1003, ¶138.

FIG. 17A



Ex. 1005, Fig. 17A (annotated).

To the extent not disclosed, it would have been obvious to have a processing component coupled to the first microphone signal and the second microphone signal. Kanamori, for example, discloses a signal delaying section 111 that can have a value anywhere from 0 to $2d/c$, which affects the orientation of the directivity pattern formed by m1 and m2. Ex. 1005, ¶¶[0175]-[0176]. It would have been obvious to use a processor to make the delaying section 111 adjustable, making the microphone device better suited to handling noise at a wider range of angles. Ex. 1003, ¶139. Moreover, processors were conventionally used to perform signal processing involving microphone arrays, including using a processor to form virtual microphones. Ex. 1003, ¶139; *e.g.*, Ex. 1013 (Elko '844), 11:43-63, 15:7-13.

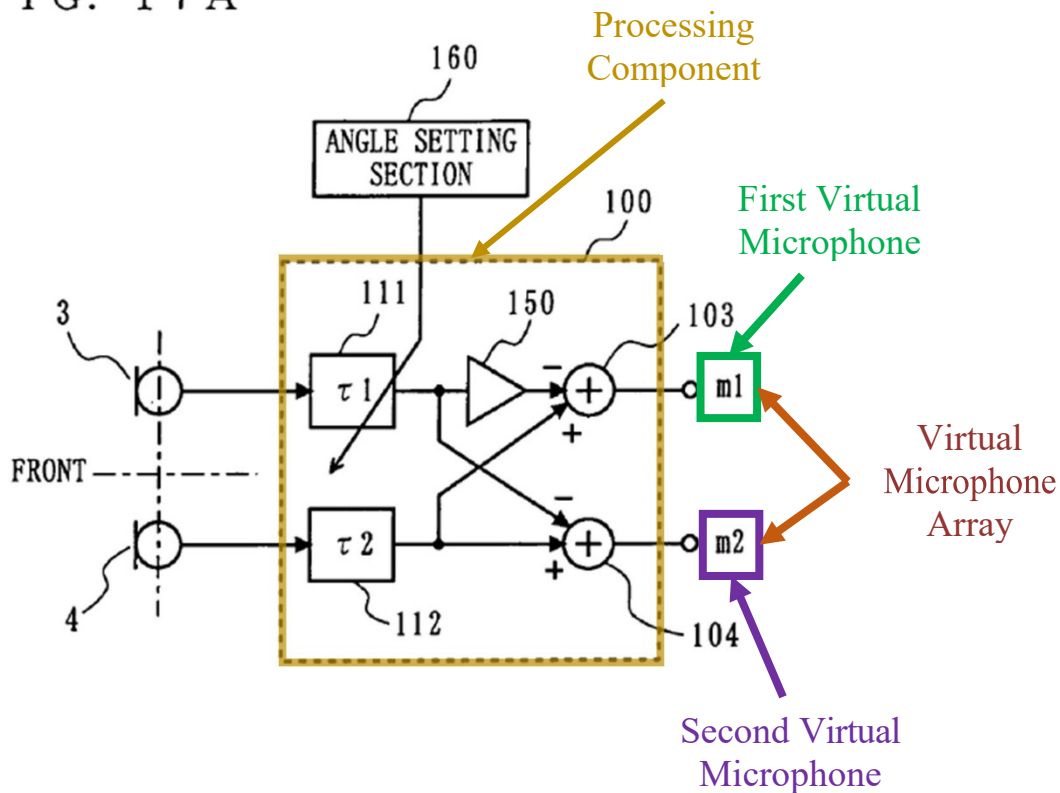
5. [29D]: “the processing component generating a virtual microphone array comprising a first virtual microphone and a second virtual microphone,”

Kanamori discloses or renders obvious [29D].

As explained for [29C], Kanamori's **directivity combining section 100** discloses or renders obvious a processing component. *Supra* §III.E.4. This **directivity combining section** performs a “directivity combining process” where the device “obtain[s] directivity by combining signals output from” “a plurality of microphone units having non-directivity.” Ex. 1005, ¶[0151]. This directivity-combining process forms **main signal m1** (first virtual microphone) and **noise reference signal m2** (second virtual microphone). Ex. 1005, ¶¶[0020], [0025], [0172]; Ex. 1003, ¶140.

Main signal $m1$ and noise reference signal $m2$ form a virtual microphone array. Ex. 1003, ¶140.

FIG. 17A



Ex. 1005, Fig. 17A (annotated).

6. [29E]: “the first virtual microphone having a first linear response to speech and a first linear response to noise,”

Kanamori discloses or suggests [29E] for the reasons explained for [1B]. *Supra* §III.C.3; Ex. 1003, ¶141.

7. **[29F]: “the first linear response to speech being substantially similar across a plurality of frequencies for a speech source at an angle relative to an axis of the microphone array,”**

Kanamori discloses or suggests [29F] for the reasons explained for [1C]. *Supra* §III.C.4; Ex. 1003, ¶142.

8. **[29G]: “the second virtual microphone having a second linear response to speech and a second linear response to noise,”**

Kanamori discloses or suggests [29G] for the reasons explained for [1F]. *Supra* §III.C.7; Ex. 1003, ¶143.

9. **[29H]: “the second linear response to noise being substantially similar to the first linear response to noise, and the second linear response to speech being substantially dissimilar to the first linear response to speech,”**

Kanamori and McCowan render obvious [29H] for the reasons explained for [1G]. *Supra* §III.C.8; Ex. 1003, ¶144.

10. **[29I]: “wherein at least one of the first linear response to noise and the second linear response to noise is non-zero in a direction towards a source of noise”**

Kanamori discloses or suggests [29I] for the reasons explained for [1H]. *Supra* §III.C.9; Ex. 1003, ¶145.

11. **[29J]: “wherein the first virtual microphone comprises the second microphone signal subtracted from a delayed version of the first microphone signal,**

Kanamori discloses or suggests [29J] for the reasons explained in claim 21.

Supra §III.D.17; Ex. 1003, ¶146.

12. **[29K]: “wherein the second virtual microphone comprises a delayed version of the first microphone signal subtracted from the second microphone signal.”**

Kanamori renders obvious [29K] for the reasons explained in claim 22. *Supra* §III.D.18; Ex. 1003, ¶147.

F. Dependent Claims 30-40

1. **Claim 30: “The device of claim 29, wherein the first linear response to speech is devoid of a null, wherein the speech is human speech.”**

Kanamori discloses or suggests this additional limitation for the reasons explained in claim 3. *Supra* §III.D.2; Ex. 1003, ¶148.

2. **Claim 31: “The device of claim 30, wherein the second linear response to speech of the second virtual microphone includes a single null oriented in a direction toward a source of the speech.”**

Kanamori and McCowan render obvious this additional limitation for the reasons explained in claim 4. *Supra* §III.D.3.a; Ex. 1003, ¶149.

3. **Claim 32: “The device of claim 31, wherein the single null is a region of the second linear response to speech having a measured response level that is lower than the measured response level of any other region of the second linear response to speech.”**

Kanamori and McCowan render obvious this additional limitation for the reasons explained in claim 5. *Supra* §III.D.3.b; Ex. 1003, ¶150.

4. **Claim 33: “The device of claim 31, wherein the second linear response to speech includes a primary lobe oriented in a direction away from the source of the speech.”**

Kanamori and McCowan render obvious this additional limitation for the reasons explained in claim 6. *Supra* §III.D.3.c; Ex. 1003, ¶151.

5. **Claim 34: “The device of claim 33, wherein the primary lobe is a region of the second linear response to speech having a measured response level that is greater than the measured response level of any other region of the second linear response to speech.”**

Kanamori and McCowan render obvious this additional limitation for the reasons explained in claim 7. *Supra* §III.D.3.d; Ex. 1003, ¶152.

6. **Claim 35: “The device of claim 31, wherein the first physical microphone and the second physical microphone are positioned along an axis and separated by a first distance.”**

Kanamori discloses or suggests this additional limitation for the reasons explained in claim 8. *Supra* §III.D.4; Ex. 1003, ¶153.

7. **Claim 36: “The device of claim 35, wherein a midpoint of the axis is a second distance from the speech source that generates the speech, wherein the speech source is located in a direction defined by an angle relative to the midpoint.”**

Kanamori discloses or suggests this additional limitation for the reasons explained in claim 9. *Supra* §III.D.5; Ex. 1003, ¶154.

8. **Claim 37: “The device of claim 36, wherein one or more of the first microphone signal and the second microphone signal is delayed.”**

Kanamori discloses or suggests this additional limitation for the reasons explained in claim 11. *Supra* §III.D.7; Ex. 1003, ¶155.

9. **Claim 38: “The device of claim 37, wherein the delay is raised to a power that is proportional to a time difference between arrival of the speech at the first virtual microphone and arrival of the speech at the second virtual microphone.”**

Kanamori discloses, or Kanamori and McCowan render obvious, this additional limitation for the reasons explained in claim 12. *Supra* §III.D.8; Ex. 1003, ¶156.

10. **Claim 39: “The device of claim 38, wherein the power is proportional to a sampling frequency multiplied by a quantity equal to a third distance subtracted from a fourth distance, the third distance being between the first physical microphone and the speech source and the fourth distance being between the second physical microphone and the speech source.”**

Kanamori discloses or Kanamori and McCowan render obvious this additional limitation for the reasons explained in claim 18. *Supra* §III.D.14; Ex. 1003, ¶157.

11. **Claim 40: “The device of claim 36, wherein one or more of the first microphone signal and the second microphone signal is multiplied by a gain factor.”**

Kanamori discloses or suggests that the second microphone signal is multiplied by a gain factor. Kanamori discloses that “signal amplifying section 150” in Figure 17A “performs constant multiplication” of the “signal output from the third microphone unit 3” (the second microphone signal) after it has been delayed at signal delaying section 111. Ex. 1005, ¶¶[0166], [0174]. The gain factor is “set at approximately 0.85.” Ex. 1005, ¶¶[0169], [0174]; Ex. 1003, ¶158.

IV. SECONDARY CONSIDERATIONS OF NONOBVIOUSNESS

Secondary considerations should be considered but do not control the obviousness conclusion. *Newell Cos. v. Kenney Mfg. Co.*, 864 F.2d 757, 768 (Fed. Cir. 1988). Where, as here, a strong *prima facie* obviousness showing exists, even relevant secondary considerations supported by substantial evidence may not dislodge

the obviousness conclusion. *Leapfrog Enters. v. Fisher-Price, Inc.*, 485 F.3d 1157, 1162 (Fed. Cir. 2007). Petitioner is aware of no evidence supporting a claim for secondary considerations.

V. DISCRETIONARY DENIAL UNDER §314(A) IS NOT APPROPRIATE.

The Board should decline to exercise its discretion under §314(a).

A. Co-Pending Litigation (*Fintiv*)

Efficiency, fairness, and the merits support institution. *Apple Inc. v. Fintiv, Inc.*, IPR2020-00019, Paper 11 (P.T.A.B. Mar. 20, 2020) (“*Fintiv*”).

1. Factor 1: Potential Stay

On petitions by Google LLC, the Board has instituted review of eight of the nine patents asserted in the Litigation:

<i>Case No.</i>	<i>Petitioner</i>	<i>Patent Reviewed</i>	<i>Final Decision Due</i>
IPR2022-00630	Google	8,280,072	September 13, 2023
IPR2022-00623	Google	7,246,058	October 31, 2023
IPR2022-00649	Google	8,019,091	November 1, 2023
IPR2022-00797	Google	8,321,213	December 1, 2023
IPR2022-01027	Google	8,467,543	December 5, 2023
IPR2022-01059	Google	10,779,080	December 7, 2023
IPR2022-01124	Google	11,122,357	January 3, 2024
IPR2022-01060, -01061	Google	8,503,691	January 3, 2024

Petitioner has moved to stay the Litigation pending these IPRs as well as other proceedings. The Court reset the hearing on Petitioner’s motion to March 21, 2023, and has accordingly not yet ruled on the motion.

The Litigation is pending in the Northern District of California (“NDCA”), where cases are routinely stayed once IPRs have been instituted. *See, e.g., Uniloc USA Inc. v. LG Elecs., U.S.A. Inc.*, No. 18-cv-06737-JST, 2019 WL 1905161 (N.D. Cal. April 29, 2019); *Symantec Corp. v. Zscaler, Inc.*, No. 17-cv-04426-JST, 2018 WL 3539267 (N.D. Cal. Jul. 23, 2018); *Twilio, Inc. v. TeleSign Corp.*, No. 16-cv-06925-LHK, 2018 WL 1609630 (N.D. Cal. April 3, 2018); *Robert Bosch Healthcare Sys., Inc. v. Cardiocom, LLC*, No. C-14-1575 EMC, 2014 WL 3107447 (N.D. Cal. July 3, 2014). Thus, a stay is highly likely, and this factor weighs against discretionary denial.

2. Factor 2: Proximity of Trial to FWD

The NDCA has not set a trial date. The Initial Case Management Conference in the Litigation is not scheduled until February 16, 2023. Thus, this factor weighs against discretionary denial. *Global Tel*Link Corp. v. HLFIP Holding, Inc.*, IPR2021-00444, Paper 14, at 16-19 (P.T.A.B. Jul. 22, 2021) (“The lack of a scheduled trial date ... typically weighs against exercising discretion to deny the Petition.”); *GAF Materials LLC v. Kirsch Research and Dev., LLC*, IPR2021-00192,

Paper 14, at 12-13 (P.T.A.B. May 25, 2021); *Skechers U.S.A., Inc. v. Nike, Inc.*, IPR2021-00160, Paper 10, at 9-11 (P.T.A.B. May 19, 2021).

3. Factor 3: Investment in Parallel Proceeding

In the Litigation, PO and Petitioner exchanged initial disclosures, preliminary infringement and invalidity contentions, and some initial documents pursuant to the local rules of the Eastern District of Texas (“EDTX”). But those disclosures will need to be updated to comply with the local patent rules of the NDCA. In discussions with Petitioner, PO has proposed serving NDCA infringement contentions no earlier than March 2023. Moreover, the parties have taken no depositions. They have neither exchanged claim-construction proposals nor submitted any claim-construction briefs. And they have served no expert reports and filed no dispositive motions.

The NDCA has not established a case schedule. An initial case management conference is scheduled for February 16, 2023. Thus, very little will be invested in the Litigation over the next several months. Even if the NDCA imposes deadlines over the following 6 months (e.g., fact discovery, claim construction), much work will still remain, including expert reports, expert discovery, dispositive motions, pre-trial motions, and trial. This “remaining investment” significantly outweighs any investment made thus far. *Samsung Elecs. Am. Inc. v. Snik LLC*, IPR2020-01428,

Paper 10, at 11 (P.T.A.B. Mar. 9, 2021); *Huawei Techs. Co. v. WSOU Invs., LLC*, IPR2021-00226, Paper 10, at 13-14 (P.T.A.B. June 10, 2021).

4. Factor 4: Overlapping Issues

If this IPR is instituted and the Litigation is stayed, Petitioner could not pursue in the Litigation any invalidity ground raised or that could have been reasonably raised in this IPR. *Cal. Inst. of Tech. v. Broadcom Ltd.*, 25 F.4th 976, 991 (Fed. Cir. 2022). Thus, duplicative efforts or conflicting decisions are unlikely.

Petitioner hereby stipulates that, if the Litigation is not stayed, Petitioner will not argue in the Litigation invalidity of the '691 patent based on the grounds in this petition or on any ground that relies on the references cited herein. This factor weighs against discretionary denial. *Sand Revolution II, LLC v. Cont'l Intermodal Grp.-Trucking LLC*, IPR2019-01393, Paper 24, at 11-12 (P.T.A.B. June 16, 2020) (informative).

5. Factor 5: The Parties

The parties are the same, but trial in the Litigation is unlikely to start before this IPR reaches a final written decision. Thus, this factor is neutral. *Google LLC v. Jawbone Innovations, LLC*, IPR2022-00630, Paper 10, at 17 (P.T.A.B. Sept. 13, 2022).

6. Factor 6: Other Circumstances

To Petitioner’s knowledge, setting aside the instituted trials in IPR2022-01060 and -01061 that Petitioner seeks to join with this petition and the companion petition, the ’691 patent has never been compared to these references by the PTO, any court, or any jury. The petition’s merits are particularly strong, which favors institution. *Fintiv*, IPR2020-00019, Paper 11, at 18. Moreover, denying institution would negate Congress’s intent in providing a 1-year period to file petitions and would encourage forum shopping as patent owners look to shield their patents from PTAB scrutiny by seeking judges with aggressive trial schedules.

Accordingly, the *Fintiv* analysis favors institution.

B. Prior IPR Petitions (*General Plastic*)

The Board instituted trials in IPR2022-01060 and -01061 on January 3, 2023, on petitions that Google filed. Apple and Samsung jointly filed “me-too” petitions with the aim of joining the proceedings on Google’s petitions, but due to settlements, the Apple/Samsung proceedings terminated before any institution decision (IPR2022-01322, -01323). None of those petitions provides grounds for denying the present petition under *General Plastic Indus. Co. v. Canon Kabushiki Kaisha*, IPR2016-01357, Paper 19, at 15-16 (P.T.A.B. Sept. 6, 2017) (precedential).

Petitioner has not filed a follow-on petition raising new unpatentability theories only after seeing PO’s arguments or the Board’s reasoning in an earlier failed

petition. Rather, Petitioner has filed a me-too petition raising identical grounds to those in IPR2022-01060, with the goal of joining that proceeding as an understudy. The concerns of abuse underlying the *General Plastic* factors therefore do not apply here. *Cent. Sec. Grp. — Nationwide, Inc. v. Ubiquitous Connectivity, LP*, IPR2019-01609, Paper 11 at 8 (P.T.A.B. Feb. 26, 2020); *Celltrion, Inc. v. Genentech, Inc.*, IPR2018-01019, Paper 11 at 9-11 (P.T.A.B. Oct. 30, 2018).

Even if the Board were to consider the *General Plastic* factors, discretionary denial is unwarranted. Amazon and the prior petitioners are not the same party and have no significant relationship. They are not co-defendants. They are competitors accused of infringement in different actions based on sales of different products. This weighs against denial. *NetNut Ltd. v. Bright Data Ltd.*, IPR2021-00465, Paper 11, at 9 (P.T.A.B. Aug. 12, 2021) (declining to extend *General Plastic* and *Valve* to different petitioner with no relationship to previous petitioners); *Mercedes-Benz USA, LLC v. Carucel Invs. L.P.*, IPR2019-01404, Paper 12, at 11-12 (Jan. 22, 2020); *Toshiba Am. Info. Sys., Inc. v. Wallelex Microelecs. Ltd.*, IPR2018-01538, Paper 11, at 20 (Mar. 5, 2019).

The remaining factors consider timing issues, e.g., when Petitioner became aware of prior art, whether the PO or Board responded to prior petitions, and the length of time that has passed. But the successive IPR filings here were caused by

PO's successive district court filings and therefore should not be held against Petitioner. Specifically, PO sued Samsung in May 2021 (2:21-cv-00186-JRG-RSP (E.D. Tex.)). PO waited four months before suing Google and Apple (6:21-cv-00985-ADA, 6:21-cv-00984-ADA (W.D. Tex.)). PO then waited another five months, until February 28, 2022, to assert the '691 patent against Petitioner. The Board should not deny Petitioner's IPR petition merely because PO sued other defendants earlier. *Toshiba*, IPR2018-01538, Paper 11, at 21-22 (PO's complaint about multiple IPRs "is not persuasive when the volume appears to be a direct result of its own litigation activity"); *id.* at 22 (it was not unfair for a petitioner to file close to one-year deadline even when co-defendants in related litigation "with distinct interests" and "distinct litigation strategies" filed earlier).

Denial on this basis would be particularly inappropriate here. PO initially asserted over 200 claims across nine patents against Petitioner. The district court's schedule required PO to provide proposed claim constructions on October 5, 2022, and to reduce the number of asserted claims to no more than 10 claims per patent and no more than 42 total claims (across all patents) by November 23, 2022. This would have given Petitioner the opportunity to file petitions addressing only the patents and claims actually asserted in the Litigation (and to do so accounting for PO's proposed claim constructions), thereby preserving both the parties' and the Board's resources.

However, the Litigation was transferred on September 21 to the NDCA, which will set a new schedule. As a result, PO never served its proposed constructions and never reduced the number of asserted patents and claims. Consequently, Petitioner could not reduce the scope of its filings. *See Facebook, Inc. v. Windy City Innovations, LLC*, 973 F.3d 1321, 1338 (Fed. Cir. 2020) (where a plaintiff has not identified a reasonable number of asserted claims by the IPR filing deadline, a defendant may be required, as a “protective measure,” to file petitions challenging “an enormous number of asserted claims”).

Because PO’s litigation strategy—not Petitioner’s conduct—caused the filing of multiple IPRs, the Board should not deny institution under *General Plastic*.

VI. DISCRETIONARY DENIAL UNDER §325(D) IS NOT APPROPRIATE.

The Board may deny institution if the Office previously considered “the same or substantially the same prior art or arguments.” 35 U.S.C. §325(d). Here, all grounds rely on Kanamori and McCowan, neither of which the PTO considered during original examination of the ’691 patent. Nor has the PTO considered “substantially the same prior art or arguments.”

Although this petition presents the same grounds as in IPR2022-01060, Petitioner is filing this petition to join that proceeding. As spelled out in the motion for joinder, Petitioner is willing to take an understudy role in the proposed consolidated proceedings.

Thus, the Board should not deny institution under §325(d).

VII. MANDATORY NOTICES

A. Real Parties-In-Interest (37 C.F.R. §42.8(b)(1))

Amazon.com, Inc., and Amazon.com Services LLC (formerly Amazon.com Services, Inc.) are the real parties-in-interest.

B. Related Matters (37 C.F.R. §42.8(b)(2))

To the best knowledge of Petitioner, the '691 patent is or has been involved in the following district court litigations and petitions for *inter partes* review:

Name	Number	Court	Filed
<i>Jawbone Innovations, LLC v. Amazon.com, Inc.</i>	3:22-cv-06727-TLT	N.D. Cal.	Nov. 29, 2021 (transferred)
<i>Jawbone Innovations, LLC v. Apple Inc.</i>	6:21-cv-00984-ADA	W.D. Tex.	Sept. 23, 2021
<i>Jawbone Innovations, LLC v. Google LLC</i>	6:21-cv-00985-ADA	W.D. Tex.	Sept. 23, 2021
<i>Jawbone Innovations, LLC v. Samsung Elec. Co., Ltd.</i>	2:21-cv-00186-JRG-RSP	E.D. Tex.	May 27, 2021
Petition for <i>Inter Partes</i> Review [Google]	IPR2022-01060	PTAB	June 9, 2022 (trial instituted Jan. 3, 2023)
Petition for <i>Inter Partes</i> Review [Google]	IPR2022-01061	PTAB	June 9, 2022 (trial instituted Jan. 3, 2023)
Petition for <i>Inter Partes</i> Review [Samsung]	IPR2022-01322	PTAB	July 27, 2022 (terminated Jan. 11, 2023)
Petition for <i>Inter Partes</i> Review [Samsung]	IPR2022-01323	PTAB	July 27, 2022 (terminated Jan. 11, 2023)

C. Lead and Backup Counsel (37 C.F.R. §42.8(b)(3))

Amazon provides the following designation of counsel:

Lead Counsel	Backup Counsel
Colin B. Heideman (Reg. No. 61,513) 2cbh@knobbe.com BoxSEAZN2L1925LP7@knobbe.com <u>Postal and Hand-Delivery Address:</u> Knobbe, Martens, Olson & Bear, LLP 925 Fourth Avenue, Suite 2500 Seattle, WA 98104 Telephone: (206) 405-2000 Facsimile: (206) 405-2001	Joseph R. Re (Reg. No. 31,291) 2jrr@knobbe.com <u>Postal and Hand-Delivery Address:</u> Knobbe, Martens, Olson & Bear, LLP 2040 Main Street, 14th Floor Irvine, CA 92614 Telephone: (949) 760-0404 Facsimile: (949) 760-9502 Nathan D. Reeves (Reg. No. 77,806) 2ndr@knobbe.com Logan P. Young (Reg. No. 79,294) 2lpy@knobbe.com <u>Postal and Hand-Delivery Address:</u> Knobbe, Martens, Olson & Bear, LLP 925 Fourth Avenue, Suite 2500 Seattle, WA 98104 Telephone: (206) 405-2000 Facsimile: (206) 405-2001

Pursuant to 37 C.F.R. §42.10(b), a Power of Attorney accompanies this Petition. Lead and backup counsel are registered practitioners associated with Customer No. 20,995 listed in that Power of Attorney.

D. Service Information (37 C.F.R. §42.8(b)(4))

Service information for counsel is provided above. Amazon consents to service by email at: BoxSEAZN2L1925LP7@knobbe.com.

E. Payment of Fees (37 C.F.R. §42.103)

The fee set forth in 37 C.F.R. §42.15(a) for this petition has been paid. Any additional fees due in connection with this petition may be charged to Deposit Account 11-1410.

F. Grounds for Standing (37 C.F.R. §42.104(a))

Amazon certifies that the '691 patent is available for *inter partes* review and that Amazon is not barred or estopped from requesting review. This petition is being filed within one year of service of the Complaint in the parties' related district-court litigation.

VIII. CONCLUSION

Petitioner respectfully requests institution of *inter partes* review and cancellation of claims 1-22 and 29-40 of the '691 patent.

Respectfully submitted,

KNOBBE, MARTENS, OLSON & BEAR, LLP

February 3, 2023

/Colin B. Heideman/

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Amazon.com, Inc. v. Jawbone Innovations, LLC
IPR Petition – U.S. Patent 8,503,691

Counsel for Petitioner,
AMAZON.COM, INC.

CERTIFICATE OF COMPLIANCE

Pursuant to 37 C.F.R. §42.24(d), the undersigned certifies that the foregoing
PETITION FOR INTER PARTES REVIEW OF CLAIMS 1-22 AND 29-40 OF
U.S. PATENT NO. 8,503,691, exclusive of the parts exempted as provided in 37
C.F.R. §42.24(a), contains 13,971 words and therefore complies with the type-vol-
ume limitations of 37 C.F.R. §42.24(a).

Respectfully submitted,

KNOBBE, MARTENS, OLSON & BEAR, LLP

February 3, 2023

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Joseph R. Re (Reg. No. 31,291)

Nathan D. Reeves (Reg. No. 77,806)

Logan P. Young (Reg. No. 79,294)

Counsel for Petitioner,
AMAZON.COM, INC.

CERTIFICATE OF SERVICE

I hereby certify that a true and correct copy of the foregoing **PETITION FOR INTER PARTES REVIEW OF CLAIMS 1-22 AND 29-40 OF U.S. PATENT NO. 8,503,691** and **EXHIBITS 1001-1019** are being served on February 3, 2023, via Federal Express overnight mail on counsel of record for U.S. Patent No. 8,503,691 as addressed below:

Mark Leonardo
Nutter McClennen & Fish LLP
Seaport West
155 Seaport Blvd.
Boston, MA 02210-2604

Courtesy copies are also being served on counsel for the patent holder in the pending litigation *Jawbone Innovations, LLC v. Amazon.com, Inc. and Amazon.com Services, Inc.*, No. 3:22-cv-06727-TLT (N.D. Cal.):

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Amazon.com, Inc. v. Jawbone Innovations, LLC
IPR Petition – U.S. Patent 8,503,691

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February 3, 2023

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