

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

AMAZON.COM, INC.,
Petitioner,

v.

JAWBONE INNOVATIONS, LLC,
Patent Owner.

IPR2023-00285
Patent 8,326,611 B2

Before GEORGIANNA W. BRADEN, NORMAN H. BEAMER, and
STEVEN M. AMUNDSON, *Administrative Patent Judges*.

AMUNDSON, *Administrative Patent Judge*.

DECISION
Granting Institution of *Inter Partes* Review
35 U.S.C. § 314
Denying Motion for Joinder as Moot
35 U.S.C. § 315(c); 37 C.F.R. § 42.122

I. INTRODUCTION

Amazon.com, Inc. (“Petitioner”) filed a Petition requesting an *inter partes* review of claims 29–44 in U.S. Patent No. 8,326,611 B2 (Exhibit 1001, “the ’611 patent”) under 35 U.S.C. §§ 311–319. Paper 1 (“Pet.”). Jawbone Innovations, LLC (“Patent Owner”) filed a Preliminary Response. Paper 9 (“Prelim. Resp.”).

Concurrently with the Petition, Petitioner filed a Motion for Joinder seeking joinder as a party in *Apple Inc. v. Jawbone Innovations, LLC*, IPR2022-01495 (“the Apple IPR”) and consolidation with the Apple IPR. Paper 3 (“Mot.”). Patent Owner filed an Opposition to Petitioner’s Motion for Joinder. Paper 7 (“Opp.”). And Petitioner filed a Reply on Petitioner’s Motion for Joinder. Paper 8 (“Reply”).

Under 37 C.F.R. § 42.4(a), we have authority to determine whether to institute an *inter partes* review. We may institute an *inter partes* review only if “the information presented in the petition filed under section 311 and any response filed under section 313 shows that there is a reasonable likelihood that the petitioner would prevail with respect to at least 1 of the claims challenged in the petition.” 35 U.S.C. § 314(a) (2018). The “reasonable likelihood” standard is “a higher standard than mere notice pleading” but “lower than the ‘preponderance’ standard to prevail in a final written decision.” *Hulu, LLC v. Sound View Innovations, LLC*, IPR2018-01039, Paper 29 at 13 (PTAB Dec. 20, 2019) (precedential).

We also have authority to consider Petitioner’s Motion for Joinder. *See* 35 U.S.C. § 315(c); 37 C.F.R. § 42.122 (2022).

After considering the Petition, the Preliminary Response, and the evidence of record, and for the reasons explained below, we determine

Petitioner has shown that there is a reasonable likelihood that it would prevail with respect to at least one of the challenged claims. Thus, we institute an *inter partes* review of claims 29–44 in the '611 patent on all challenges included in the Petition.

Additionally, for the reasons explained below, we deny Petitioner's Motion for Joinder as moot.

II. BACKGROUND

A. Real Parties in Interest

Petitioner identifies the following entities as the real parties in interest: Amazon.com, Inc. and Amazon.com Services LLC (formerly Amazon.com Services, Inc.). Pet. 100. Patent Owner identifies itself as the real party in interest. Paper 5, 2. The parties do not raise any issue about real parties in interest.

B. Related Matters

Petitioner and Patent Owner identify the following civil actions as related matters involving the '611 patent:

- *Jawbone Innovations, LLC v. Samsung Electronics Co., Ltd. et al.*, No. 2:21-cv-00186-JRG (E.D. Tex. filed May 27, 2021);
- *Jawbone Innovations, LLC v. Apple Inc.*, No. 6:21-cv-00984-ADA (W.D. Tex. filed Sept. 23, 2021);
- *Jawbone Innovations, LLC v. Google LLC*, No. 6:21-cv-00985-ADA (W.D. Tex. filed Sept. 23, 2021); and
- *Jawbone Innovations, LLC v. Amazon.com, Inc. et al.*, No. 5:22-cv-06727-NC (N.D. Cal. filed Nov. 29, 2021).

Pet. 100; Paper 5, 2–3.

Petitioner and Patent Owner identify the following Board proceedings as related matters involving the '611 patent:

- *Google LLC v. Jawbone Innovations, LLC*, IPR2022-00604 (PTAB filed Feb. 22, 2022);
- *Google LLC v. Jawbone Innovations, LLC*, IPR2022-00889 (PTAB filed May 16, 2022);
- *Apple Inc. v. Jawbone Innovations, LLC*, IPR2022-01085 (PTAB filed June 3, 2022); and
- *Apple Inc. v. Jawbone Innovations, LLC*, IPR2022-01495 (PTAB filed Sept. 2, 2022).

Pet. 100; Paper 5, 2.

The following Board proceeding also involves the '611 patent:
Amazon.com, Inc. v. Jawbone Innovations, LLC, IPR2023-00286 (PTAB filed Nov. 28, 2022).

C. The '611 Patent (Exhibit 1001)

The '611 patent, titled “Acoustic Voice Activity Detection (AVAD) for Electronic Systems,” issued on December 4, 2012, from an application filed on October 26, 2009. Ex. 1001, codes (22), (45), (54). The patent identifies that application as a continuation-in-part of the following applications: (1) U.S. Patent Application No. 11/805,987, filed on May 25, 2007; and (2) U.S. Patent Application No. 12/139,333, filed on June 13, 2008. *Id.* at 1:8–11, code (63). The patent claims priority to U.S. Provisional Patent Application No. 61/108,426, filed on October 24, 2008. *Id.* at 1:6–7, code (60). The patent states that the disclosure relates generally “to noise suppression” and more particularly “to noise suppression systems, devices, and methods for use in acoustic applications.” *Id.* at 1:16–19; *see id.* at code (57).

The '611 patent explains that the “ability to correctly identify voiced and unvoiced speech is critical to many speech applications including speech

recognition, speaker verification, noise suppression, and many others.”

Ex. 1001, 1:23–26. In “the speaker’s environment,” however, “there may exist one or more noise sources that pollute the speech signal, the signal of interest, with unwanted acoustic noise.” *Id.* at 1:28–31. The unwanted acoustic noise “makes it difficult or impossible for the receiver, whether human or machine, to understand the user’s speech.” *Id.* at 1:31–32.

The ’611 patent also explains that “[t]ypical methods for classifying voiced and unvoiced speech have relied mainly on the acoustic content of single microphone data, which is plagued by problems with noise and the corresponding uncertainties in signal content.” Ex. 1001, 1:33–36.

According to the patent, non-acoustic methods “have been employed successfully in commercial products,” but “an acoustic-only solution is desired in some cases (e.g., for reduced cost, as a supplement to the non-acoustic sensor, etc.).” *Id.* at 1:41–46.

Toward that end, the ’611 patent discloses “Acoustic Voice Activity Detection (AVAD) methods and systems” that use physical microphones “to generate virtual directional microphones which have very similar noise responses and very dissimilar speech responses.” Ex. 1001, 3:57–62, code (57); *see id.* at 4:21–25, 5:35–52, 17:43–45. “The ratio of the energies of the virtual microphones is then calculated over a given window size and the ratio can then be used with a variety of methods to generate a VAD signal.” *Id.* at 3:62–65, code (57). “The virtual microphones can be constructed using either a fixed or an adaptive filter.” *Id.* at 3:65–66, code (57); *see id.* at 4:27–30.

The ’611 patent defines various terms. Ex. 1001, 17:57–18:32. The patent includes the following definitions:

- The term “speech” means desired speech of the user.
- The term “noise” means unwanted environmental acoustic noise.
- The term “O₁” means a first physical omnidirectional microphone used to form a microphone array.
- The term “O₂” means a second physical omnidirectional microphone used to form a microphone array.
- The term “virtual microphones (VM)” or “virtual directional microphones” means a microphone constructed using two or more omnidirectional microphones and associated signal processing.
- The term “V₁” means the virtual directional “speech” microphone, which has no nulls.
- The term “V₂” means the virtual directional “noise” microphone, which has a null for the user’s speech.

Id. at 18:10–32.

The ’611 patent explains that “V₂ is configured in such a way that it has minimal response to the speech of the user” and “V₁ is configured so that it does respond to the user’s speech but has a very similar noise magnitude response to V₂.” Ex. 1001, 4:21–25. “A further refinement is the use of an adaptive filter to further minimize the speech response of V₂.” *Id.* at 4:27–28; *see id.* at 6:10–11.

The '611 patent's Figure 3 (reproduced below) depicts a block diagram of a virtual microphone V_2 :

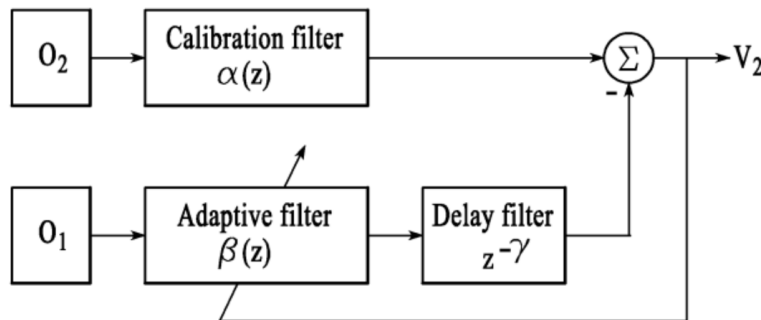


FIG.3

Figure 3 illustrates the following components where “z” denotes a discrete frequency domain and “ γ ” is a fixed delay that depends on the size of the microphone array:

- (1) a physical omnidirectional microphone O_1 providing a signal to an adaptive filter $\beta(z)$;
- (2) the adaptive filter $\beta(z)$ providing a signal to a delay filter $z^{-\gamma}$;
- (3) a physical omnidirectional microphone O_2 providing a signal to a calibration filter $\alpha(z)$;
- (4) a summing circuit (Σ) summing (i) a positive signal from the calibration filter $\alpha(z)$ and (ii) a negative signal from the delay filter $z^{-\gamma}$ to form a virtual microphone V_2 ; and
- (5) the virtual microphone V_2 providing a signal to the adaptive filter $\beta(z)$.

See Ex. 1001, 1:62–63, 4:41, 5:22–33, 6:1–16, Fig. 3.

The '611 patent's Figure 4 (reproduced below) depicts a block diagram of a virtual microphone V_1 :

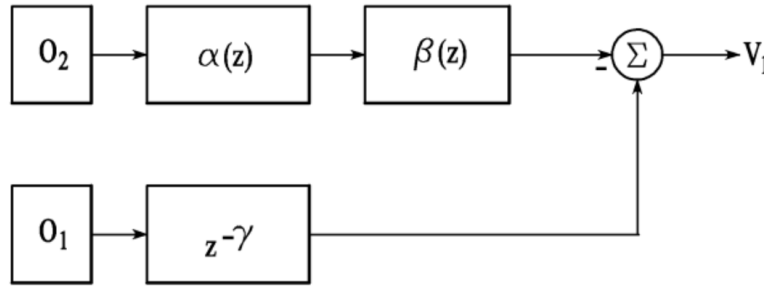


FIG.4

Figure 4 illustrates the following components where “z” denotes a discrete frequency domain and “ γ ” is a fixed delay that depends on the size of the microphone array:

- (1) a physical omnidirectional microphone O_1 providing a signal to a delay filter $z^{-\gamma}$;
- (2) a physical omnidirectional microphone O_2 providing a signal to a calibration filter $\alpha(z)$;
- (3) the calibration filter $\alpha(z)$ providing a signal to a filter $\beta(z)$; and
- (4) a summing circuit (Σ) summing (i) a positive signal from the delay filter $z^{-\gamma}$ and (ii) a negative signal from the filter $\beta(z)$ to form a virtual microphone V_1 .

See Ex. 1001, 1:64–65, 4:41, 5:22–33, 6:17–19, Fig. 4.

The '611 patent provides the following equations for virtual microphones V_1 and V_2 constructed using physical omnidirectional microphones O_1 and O_2 :

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

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

Ex. 1001, 5:20–28. The patent explains that in these equations (1) the filter $\alpha(z)$ “compensate[s] O_2 ’s response so that it is the same as O_1 ” and (2) the filter $\beta(z)$ “describes the relationship between O_1 and calibrated O_2 for speech.” *Id.* at 5:29–32; *see id.* at 23:11–41. The patent also explains that by “varying the magnitude and sign of the delays and gains, a wide variety of virtual microphones (VMs), also referred to herein as virtual directional microphones, can be realized.” *Id.* at 21:7–10.

The ’611 patent discloses an adaptive filter $\beta(z)$ that “minimize[s] the output of V_2 when only speech is being received by O_1 and O_2 ” and preferably has the following construction:

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

Ex. 1001, 6:1–11, 6:35–38. “Any adaptive process may be used,” such as “a normalized least-mean squares (NLMS) algorithm.” *Id.* at 6:14–15.

With $\beta(z)$ suitably selected, “the ratio for speech should be relatively high (e.g., greater than approximately 2) and the ratio for noise should be relatively low (e.g., less than approximately 1.1).” Ex. 1001, 6:29–32. “The ratio calculated will depend on both the relative energies of the speech and noise as well as the orientation of the noise and the reverberance of the environment.” *Id.* at 6:32–35.

The '611 patent's Figure 5 (reproduced below) depicts a flow diagram of acoustic voice activity detection:

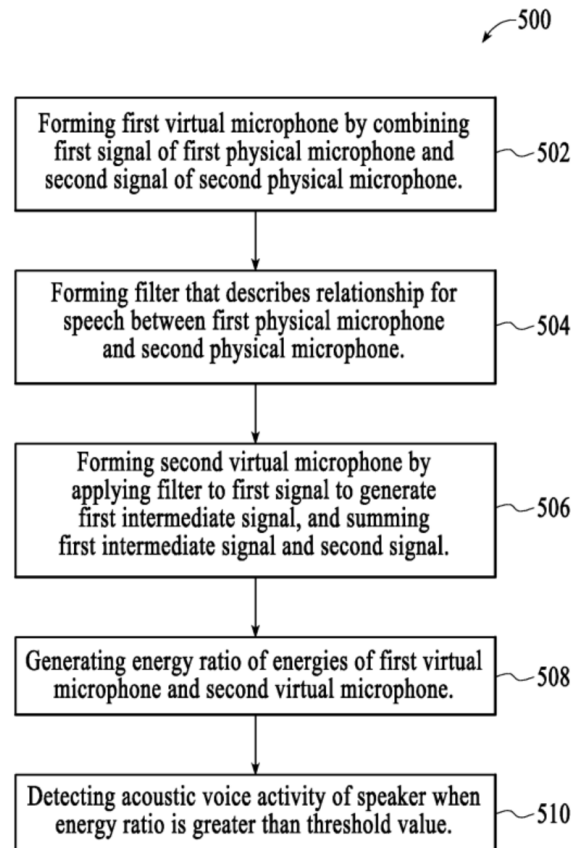


FIG.5

Figure 5 illustrates “acoustic voice activity detection 500” including steps 502 through 510. Ex. 1001, 1:66–67, 6:60–61, Fig. 5.

At step 502, the “detection comprises forming a first virtual microphone by combining a first signal of a first physical microphone and a second signal of a second physical microphone.” Ex. 1001, 6:61–64, Fig. 5. At step 504, the “detection comprises forming a filter that describes a relationship for speech between the first physical microphone and the second physical microphone.” *Id.* at 6:64–67, Fig. 5. At step 506, the “detection comprises forming a second virtual microphone by applying the filter to the

first signal to generate a first intermediate signal, and summing the first intermediate signal and the second signal.” *Id.* at 6:67–7:3, Fig. 5. At step 508, the “detection comprises generating an energy ratio of energies of the first virtual microphone and the second virtual microphone.” *Id.* at 7:3–5, Fig. 5. At step 510, the “detection comprises detecting acoustic voice activity of a speaker when the energy ratio is greater than a threshold value.” *Id.* at 7:5–7, Fig. 5.

D. The Challenged Claims

Petitioner challenges independent method claim 29, claims 30–43 that depend directly or indirectly from claim 29, and independent method claim 44. Pet. 4, 10–91.

Claims 29 and 44 exemplify the challenged claims and read as follows (with numbers and letters added for reference purposes):¹

29. [29pre] A method comprising:

[29a] forming a first virtual microphone;

[29b] forming a filter by generating a first quantity by applying a calibration to a second signal of a second physical microphone, generating a second quantity by applying the delay to a first signal of a first physical microphone, and forming the filter as a ratio of the first quantity to the second quantity;

[29c] forming a second virtual microphone by applying the filter to the first signal to generate a first intermediate signal, and summing the first intermediate signal and the second signal; and

¹ We use the same numbers and letters that Petitioner uses to identify the claim language. *See* Pet. 1, 3 (Listing of Challenged Claims). These annotations do not impact our analysis.

[29d] generating a ratio of energies of the first virtual microphone and the second virtual microphone and detecting acoustic voice activity using the ratio.

44. [44pre] A method comprising:

[44a] forming a first virtual microphone by generating a first combination of a first signal and a second signal, wherein the first signal is received from a first physical microphone and the second signal is received from a second physical microphone;

[44b] forming a filter by generating a first quantity by applying a calibration to at least one of the first signal and the second signal, generating a second quantity by applying a delay to the first signal, and forming the filter as a ratio of the first quantity to the second quantity; and

[44c] forming a second virtual microphone by applying the filter to the first signal to generate a first intermediate signal and summing the first intermediate signal and the second signal; and

[44d] determining a presence of acoustic voice activity of a speaker when an energy ratio of energies of the first virtual microphone and the second virtual microphone is greater than a threshold value.

Ex. 1001, 38:33–47, 40:3–21.

E. The Asserted References

For its challenges, Petitioner relies on the following references:

Name	Reference	Exhibit
Avendano '880	US 8,194,880 B2, issued June 5, 2012 (based on an application filed January 29, 2007)	1005
Avendano '252	US 8,204,252 B1, issued June 19, 2012 (based on an application filed March 31, 2008)	1006
Hou	US 7,155,019 B2, issued December 26, 2006 (based on an application filed March 14, 2001)	1008

Pet. 4–5, 10–91.

Petitioner asserts that Avendano ’880, Avendano ’252, and Hou qualify as prior art under § 102(e). Pet. 5–6; *see* 35 U.S.C. § 102(e) (2006).²

At this stage of the proceeding, Patent Owner does not dispute that each reference qualifies as prior art. *See, e.g.*, Prelim. Resp. 3, 5–10.

F. The Asserted Challenges to Patentability

Petitioner asserts the following challenges to patentability:

Claim(s) Challenged	35 U.S.C. §	Reference(s)/Basis
29–31, 33–37, 41–44	103(a)	Avendano ’880, Hou
32, 38–40	103(a)	Avendano ’880, Avendano ’252, Hou

Pet. 4, 10–91.

G. Testimonial Evidence

To support its challenges, Petitioner relies on the declaration of Thomas W. Kenny, Ph.D. (Exhibit 1003). Dr. Kenny states, “I received a B.S. in Physics from University of Minnesota, Minneapolis in 1983, and a Ph.D. in Physics from University of California at Berkeley in 1989” and “was educated as a Physicist specializing in sensors and measurement.” Ex. 1003 ¶ 3. Dr. Kenny also states, “I have been retained” to “offer technical opinions” about the ’611 patent and “prior art references relating to its subject matter.” *Id.* ¶ 16.

² The Leahy-Smith America Invents Act (“AIA”), Pub. L. No. 112-29, 125 Stat. 284 (2011), amended 35 U.S.C. § 102 and § 103 effective March 16, 2013. Because the filing date of the challenged claims predates the AIA’s amendments to § 102 and § 103, this decision refers to the pre-AIA versions of § 102 and § 103.

III. MOTION FOR JOINDER

In November 2022, concurrently with the Petition, Petitioner filed a Motion for Joinder seeking joinder as a party in the Apple IPR and consolidation with the Apple IPR. Mot. 1.

In December 2022, Patent Owner filed an Opposition to Petitioner's Motion for Joinder, requesting that the Board deny the joinder motion because "Apple and Patent Owner have recently settled their dispute and will seek to file a motion to terminate the Apple IPR." Opp. 1.

In January 2023, the Board terminated the Apple IPR in view of representations by Apple and Patent Owner that they had "reached settlements as to all the disputes in this proceeding and as to the '611 patent." IPR2022-01495, Paper 6 at 2; Paper 8 at 2–3.

Later in January 2023, Petitioner filed a Reply on Petitioner's Motion for Joinder, stating that the "joinder motion should be deemed withdrawn or denied as moot." Reply 1.

Accordingly, we deny Petitioner's Motion for Joinder as moot.

IV. PATENTABILITY ANALYSIS

A. Legal Principles: Obviousness

A patent may not be obtained "if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains." 35 U.S.C. § 103(a) (2006). An obviousness analysis involves underlying factual inquiries including (1) the scope and content of the prior art; (2) differences between the claimed invention and the prior art; (3) the level of ordinary skill in the art; and (4) where in evidence, objective indicia of

nonobviousness, such as commercial success, long-felt but unsolved needs, and failure of others.³ *Graham v. John Deere Co.*, 383 U.S. 1, 17–18, 35–36 (1966); *Apple Inc. v. Samsung Elecs. Co.*, 839 F.3d 1034, 1047–48 (Fed. Cir. 2016) (en banc). When evaluating a combination of references, an obviousness analysis should address “whether there was an apparent reason to combine the known elements in the fashion claimed by the patent at issue.” *KSR Int’l Co. v. Teleflex Inc.*, 550 U.S. 398, 418 (2007).

We analyze the obviousness issues according to these principles.

B. Level of Ordinary Skill in the Art

Factors pertinent to determining the level of ordinary skill in the art include (1) the educational level of the inventor; (2) the type of problems encountered in the art; (3) prior-art solutions to those problems; (4) the rapidity with which innovations are made; (5) the sophistication of the technology; and (6) the educational level of workers active in the field. *Envtl. Designs, Ltd. v. Union Oil Co.*, 713 F.2d 693, 696–97 (Fed. Cir. 1983). Not all factors may exist in every case, and one or more of these or other factors may predominate in a particular case. *Id.* These factors are not exhaustive, but merely a guide to determining the level of ordinary skill in the art. *Daiichi Sankyo Co. v. Apotex, Inc.*, 501 F.3d 1254, 1256 (Fed. Cir. 2007). Moreover, the prior art itself may reflect an appropriate skill level. *Okajima v. Bourdeau*, 261 F.3d 1350, 1355 (Fed. Cir. 2001).

Petitioner asserts that a person of ordinary skill in the art at the time of the alleged invention would have had “at least a bachelor of science in electrical engineering, computer engineering, computer science, mechanical

³ The record does not include evidence or argument regarding objective indicia of nonobviousness.

engineering, or a related discipline, with at least two years of relevant experience in a field related to acoustics, speech recognition, speech detection, or signal processing.” Pet. 10. Petitioner also asserts that “[a]dditional education or industry experience may compensate for a deficit in the other.” *Id.* Dr. Kenny’s testimony supports Petitioner’s assertions. *See* Ex. 1003 ¶¶ 22–23.

For purposes of the Preliminary Response, Patent Owner “utilizes Petitioner’s proposed level of skill in the art.” Prelim. Resp. 4 (quoting Pet. 10).

Based on the current record and for purposes of analysis, we accept Petitioner’s description of an ordinarily skilled artisan as consistent with the ’611 patent and the asserted prior art.

C. Claim Construction

We construe claim terms “using the same claim construction standard” that district courts use to construe claim terms in civil actions under 35 U.S.C. § 282(b). *See* 37 C.F.R. § 42.100(b). Under that standard, claim terms “are given their ordinary and customary meaning, which is the meaning the term would have to a person of ordinary skill in the art at the time of the invention.” *Power Integrations, Inc. v. Fairchild Semiconductor Int’l, Inc.*, 904 F.3d 965, 971 (Fed. Cir. 2018) (citing *Phillips v. AWH Corp.*, 415 F.3d 1303, 1312–13 (Fed. Cir. 2005) (en banc)). The meaning of claim terms may be determined by “look[ing] principally to the intrinsic evidence of record, examining the claim language itself, the written description, and the prosecution history, if in evidence.” *DePuy Spine, Inc. v. Medtronic Sofamor Danek, Inc.*, 469 F.3d 1005, 1014 (Fed. Cir. 2006) (citing *Phillips*, 415 F.3d at 1312–17).

Petitioner contends that “[n]o formal claim constructions are necessary.” Pet. 10.

Patent Owner “believes that claim construction is not required to resolve any issues.” Prelim. Resp. 3.

Based on the current record, we determine that no claim term requires an explicit construction to decide whether Petitioner satisfies the “reasonable likelihood” standard for instituting trial. “[O]nly those terms need be construed that are in controversy, and only to the extent necessary to resolve the controversy.” *Vivid Techs., Inc. v. Am. Sci. & Eng’g, Inc.*, 200 F.3d 795, 803 (Fed. Cir. 1999); see *Nidec Motor Corp. v. Zhongshan Broad Ocean Motor Co.*, 868 F.3d 1013, 1017 (Fed. Cir. 2017).

*D. Alleged Obviousness over Avendano ’880
and Hou: Claims 29–31, 33–37, and 41–44*

Petitioner contends that claims 29–31, 33–37, and 41–44 are unpatentable under § 103(a) as obvious over Avendano ’880 and Hou. See Pet. 4, 10–70. Patent Owner disputes Petitioner’s contentions. See Prelim. Resp. 5–10. Below, we provide overviews of Avendano ’880 and Hou, and then we consider the obviousness issues. As explained below, Petitioner establishes sufficiently for purposes of institution that the combined disclosures in Avendano ’880 and Hou teach the subject matter in claims 29–31, 33–37, and 41–44.

1. OVERVIEW OF AVENDANO ’880 (EXHIBIT 1005)

Avendano ’880 is a U.S. patent titled “System and Method for Utilizing Omni-Directional Microphones for Speech Enhancement,” filed on January 29, 2007, and issued on June 5, 2012. Ex. 1005, codes (12), (22),

(45), (54). Avendano '880 states that the invention “relates generally to audio processing.” *Id.* at 1:24; *see id.* at code (57).

Avendano '880 discloses systems and methods “for utilizing inter-microphone level differences (ILD) to attenuate noise and enhance speech.” Ex. 1005, 2:5–7, 3:13–17, code (57). Preferably, “the ILD is based on energy level differences of a pair of omni-directional microphones.” *Id.* at 2:7–9, code (57). For example, “a primary acoustic signal is received by a primary microphone, and a secondary acoustic signal is received by a secondary microphone.” *Id.* at 2:14–17, code (57). “The primary and secondary acoustic signals are converted into primary and secondary electric signals for processing.” *Id.* at 2:17–19, code (57).

“A differential microphone array (DMA) module processes the primary and secondary electric signals to determine a cardioid primary signal and a cardioid secondary signal.” Ex. 1005, 2:20–22, code (57). Preferably, “the primary and secondary electric signals are delayed by a delay node.” *Id.* at 2:22–24. “The cardioid primary signal is then determined by taking a difference between the primary electric signal and the delayed secondary electric signal, while the cardioid secondary signal is determined by taking a difference between the secondary electric signal and the delayed primary electric signal.” *Id.* at 2:24–29.

“The cardioid signals are filtered through a frequency analysis module” that “takes the signals and mimics the frequency analysis of the cochlea (i.e., cochlear domain).” Ex. 1005, 2:34–36, code (57). “Energy levels associated with the cardioid primary signal and the cardioid secondary signals are then computed (e.g., as power estimates) and the results are processed by an ILD module” to obtain the ILD. *Id.* at 2:41–45, code (57).

The ILD module may obtain the ILD by “using a non-linear combination,” e.g., by “dividing the power estimate associated with the cardioid primary signal by the power estimate associated with the cardioid secondary signal.” *Id.* at 2:44–48, code (57). The ILD “may then be used” to (1) “suppress unwanted sound sources” and (2) “enhance the speech of the primary acoustic signal.” *Id.* at 2:48–50, code (57).

Avendano '880's Figure 1a (reproduced below) depicts an environment for practicing embodiments of the invention:

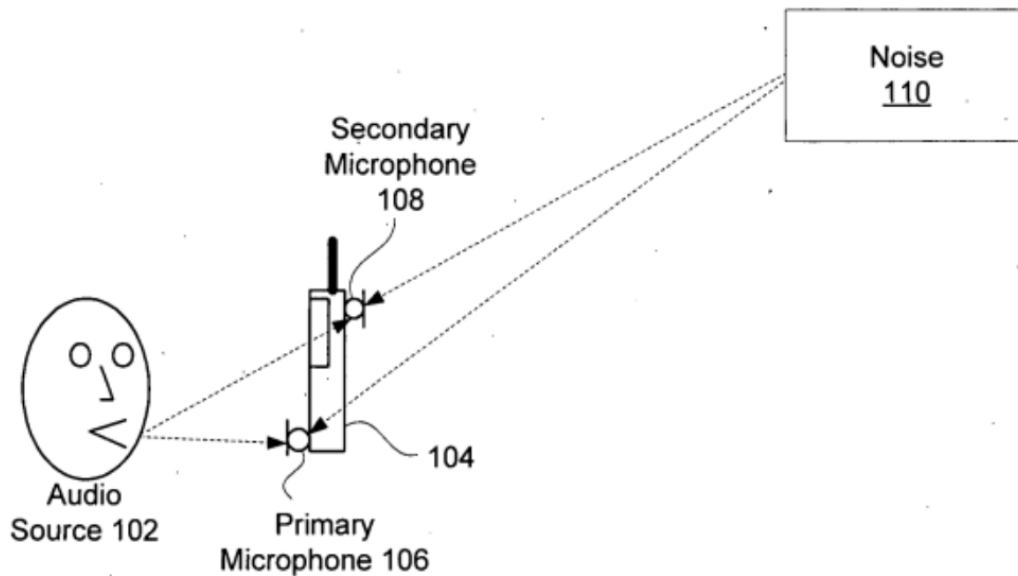


FIG. 1a

Figure 1a illustrates a user who “provides an audio (speech) source 102 to an audio device 104” comprising two microphones: “a primary microphone 106 relative to the audio source 102 and a secondary microphone 108 located a distance, d , away from the primary microphone 106.” Ex. 1005, 3:27–33, Fig. 1a. Preferably, “the microphones 106 and 108 are omni-directional microphones.” *Id.* at 3:33–35.

“While the microphones 106 and 108 receive sound (i.e., acoustic signals) from the audio source 102, the microphones 106 and 108 also pick up noise 110.” Ex. 1005, 3:36–38, Fig. 1a. Noise 110 “may comprise any sounds from one or more locations different than the audio source 102” and “may include reverberations and echoes.” *Id.* at 3:38–42.

Avendano ’880’s Figure 2 (reproduced below) depicts an exemplary audio device:

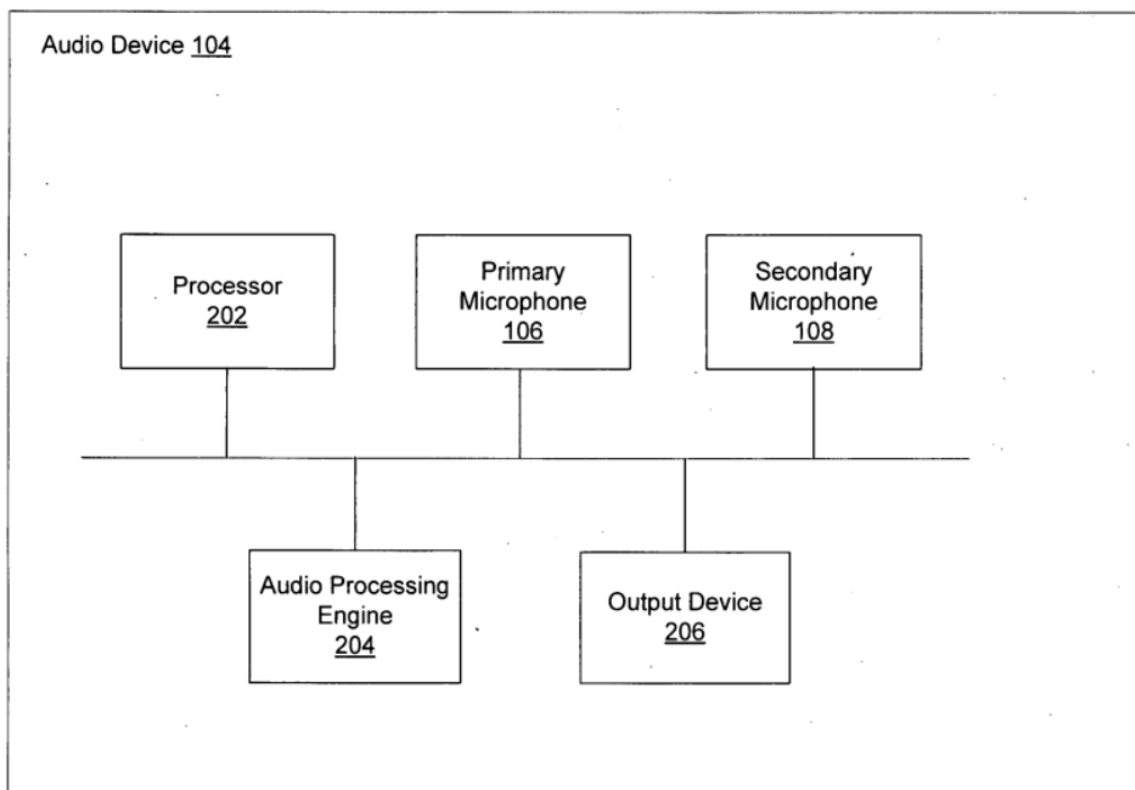


FIG. 2

Figure 2 illustrates audio device 104 including primary microphone 106, secondary microphone 108, processor 202, audio processing engine 204, and output device 206. Ex. 1005, 3:61–66, Fig. 2; *see id.* at 2:57–58. To permit processing by audio processing engine 204, an acoustic signal received by primary microphone 106 (a primary acoustic signal X_1) is converted into a

primary electric signal, and an acoustic signal received by secondary microphone 108 (a secondary acoustic signal X_2) is converted into a secondary electric signal. *Id.* at 4:6–8, 4:23–27. Audio processing engine 204 “is embodied within a memory device.” *Id.* at 4:22–23.

Avendano ’880’s Figure 4a (reproduced below) depicts an embodiment of audio processing engine 204:

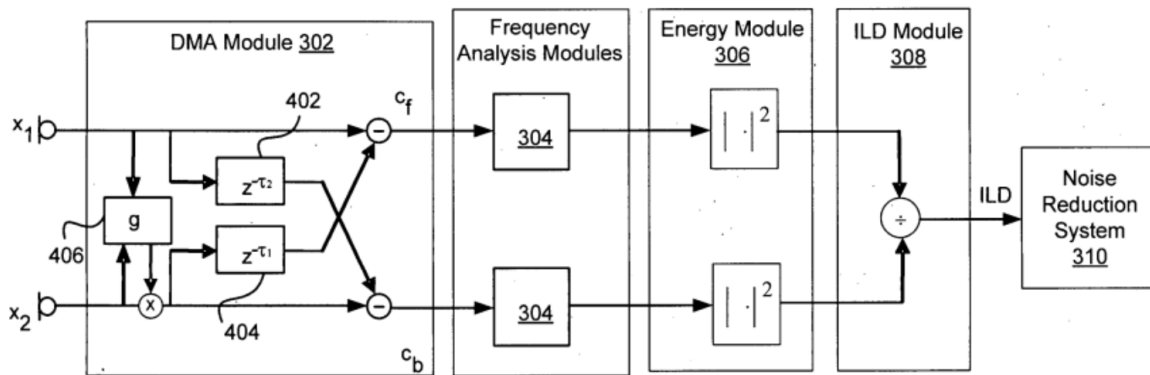


FIG. 4a

Figure 4a illustrates one implementation of audio processing engine 204 including differential microphone array (DMA) module 302, frequency-analysis module 304, energy module 306, ILD module 308, and noise-reduction system 310. Ex. 1005, 2:61–63, 5:15–17, 6:36–40, Fig. 4a; *see id.* at 2:20. As Figure 4a shows, DMA module 302 receives an electric signal x_1 corresponding to the primary acoustic signal X_1 and an electric signal x_2 corresponding to the secondary acoustic signal X_2 . *See id.* at 4:6–8, 4:23–27, 5:17–20, Fig. 4a.

After receiving the electric signals x_1 and x_2 , DMA module 302 determines “sounds and signals in a front and back cardioid region about” audio device 104 by “delaying and subtracting the acoustic signals captured by the microphones 106 and 108.” Ex. 1005, 4:29–33. By doing so, DMA

module 302 creates “directional patterns” for the microphones. *Id.* at 4:27–29; *see id.* at 4:40–46.

For example, “one directional pattern created by the DMA module 302 allows sounds generated from an audio source 102 within a front cardioid region around the audio device 104 to be received,” and “a second pattern allows sounds from a second audio source 102 within a back cardioid region around the audio device 104 to be received.” Ex. 1005, 4:47–52. “Sounds from audio sources 102 beyond these regions may also be received but the sounds may be attenuated.” *Id.* at 4:52–54.

As Figure 4a shows, DMA module 302 delays the electric signal x_1 corresponding to the primary acoustic signal X_1 via delay node 402 with delay $z^{-\tau_2}$, and DMA module 302 delays a gain-adjusted version of the electric signal x_2 corresponding to the secondary acoustic signal X_2 via delay node 404 with delay $z^{-\tau_1}$.⁴ Ex. 1005, 5:20–24, 5:36–37, 8:52–60, Fig. 4a. DMA module 302 generates a cardioid primary signal C_f and a cardioid secondary signal C_b according to the following equations for C_f and C_b :

$$C_f = X_1 - z^{-\tau_1}gX_2 \qquad C_b = gX_2 - z^{-\tau_2}X_1$$

⁴ In contrast to Figure 4a, Avendano ’880’s written description identifies (1) the delay node for the electric signal x_1 corresponding to the primary acoustic signal X_1 as delay node 404 with delay $z^{-\tau_1}$ and (2) the delay node for the electric signal x_2 corresponding to the secondary acoustic signal X_2 as delay node 404 with delay $z^{-\tau_2}$. Ex. 1005, 5:20–24. But the written description includes equations that use the delays $z^{-\tau_1}$ and $z^{-\tau_2}$ consistently with Figure 4a, i.e., delay node 402 with delay $z^{-\tau_2}$ applied to the electric signal x_1 and delay node 404 with delay $z^{-\tau_1}$ applied to the electric signal x_2 after gain adjustment. *Id.* at 5:25–35, Fig. 4a.

where the gain factor “g” is “computed by the gain module 406 to equalize the signal levels.” *Id.* at 5:25–37; *see id.* at 8:52–60. The gain factor “g” equalizes the signal levels to prevent a “loss of performance when the microphone signals have different levels.” *Id.* at 5:37–39.

Frequency-analysis module 304 “takes the cardioid signals and mimics the frequency analysis of the cochlea (i.e., cochlear domain).” Ex. 1005, 4:55–59; *see id.* at 5:41–48, code (57). To simulate the cochlear domain, frequency-analysis module 304 may employ a filter bank or similar mechanism, e.g., a short-time Fourier transform (STFT), sub-band filter banks, or modulated complex lapped transforms. *Id.* at 2:34–41, 4:55–64, 9:67–10:3. “Because most sounds (e.g., acoustic signals) are complex and comprise more than one frequency, a sub-band analysis on the acoustic signal determines what individual frequencies are present in the complex acoustic signal during a frame (e.g., a predetermined period of time).” *Id.* at 4:64–5:1.

Next, energy module 306 “takes the signals from the frequency analysis module 304 and calculates the power estimates associated with the cardioid primary signal (C_f) and the cardioid secondary signal (C_b).” Ex. 1005, 5:49–52. For example, “the power estimates may be mathematically determined by squaring and integrating an absolute value of the output of” frequency-analysis module 304. *Id.* at 5:52–55.

Avendano ’880 discloses the following equations for calculating the energy level $E_f(t, \omega)$ associated with the primary microphone signal and the energy level $E_b(t, \omega)$ associated with the secondary microphone signal by squaring and integrating:

$$E_f(t, \omega) = \int_{\text{frame}} |C_f(t', \omega)|^2 dt'$$

$$E_b(t, \omega) = \int_{\text{frame}} |C_b(t', \omega)|^2 dt'$$

Ex. 1005, 5:58–6:7.

Energy module 306 supplies the calculated energy levels $E_f(t, \omega)$ and $E_b(t, \omega)$ to ILD module 308. Ex. 1005, 6:9–10, Fig. 4a. Then, ILD module 308 determines the ILD “in a non-linear manner by taking a ratio of the energy levels” according to the following equation:

$$\text{ILD}(t, \omega) = E_f(t, \omega) / E_b(t, \omega)$$

Id. at 6:10–13.

“Once the ILD is determined, the signals are processed through a noise reduction system 310.” Ex. 1005, 6:35–36, Fig. 4a. Noise reduction system 310 may include “a noise estimate module 312, a filter module 314, a filter smoothing module 316, a masking module 318, and a frequency synthesis module 320.” *Id.* at 6:36–40, Fig. 3. Noise estimate module 312 may calculate a noise estimate using, among other things, the value $\lambda_1(t, \omega)$ determined as follows:

$$\lambda_1(t, \omega) \approx 0 \text{ if } \text{ILD}(t, \omega) < \text{threshold}$$

$$\lambda_1(t, \omega) \approx 1 \text{ if } \text{ILD}(t, \omega) > \text{threshold}$$

Id. at 6:48–7:6.

Avendano '880's Figure 6 (reproduced below) depicts “a polar plot of a front-to-back cardioid directivity pattern and ILD diagram”:

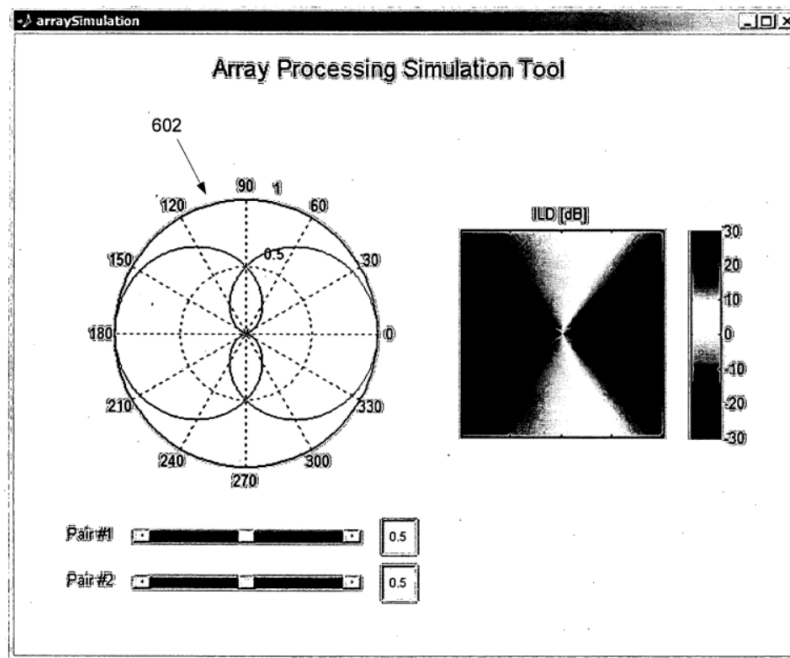


FIG. 6

Figure 6 illustrates “front-to-back cardioid directivity pattern 602” with the directivity “extended in the forward and backward directions (i.e., along the x-axis).” Ex. 1005, 9:29–37, Fig. 6; *see id.* at 3:1–3. “The extension in the forward and backward directions allows significant ILD cues to be obtained from acoustic sources further away from the microphones 106 and 108.” *Id.* at 9:37–39. “As a result, the omni-directional microphones 106 and 108 can achieve acoustic characteristics that mimic those of directional microphones.” *Id.* at 9:39–42.

2. OVERVIEW OF HOU (EXHIBIT 1008)

Hou is a U.S. patent titled “Adaptive Microphone Matching in Multi-Microphone Directional System,” filed on March 14, 2001, and issued on December 26, 2006. Ex. 1008, codes (12), (22), (45), (54). Hou states that the invention “relates to multi-microphone sound pick-up systems” and

“matching microphone sensitivity in multi-microphone sound pick-up systems.” *Id.* at 1:22–25.

Hou explains that a “multi-microphone array can selectively enhance sounds coming from certain directions while suppressing interference coming from other directions.” Ex. 1008, 1:30–32. To “achieve good directionality,” however, the sensitivities of the microphones must match. *Id.* at 1:48–50; *see id.* at 1:67–2:2. “When the sensitivities of the microphones are not properly matched, then the directionality is substantially degraded and thus the ability to suppress interference coming from a particular direction is poor.” *Id.* at 1:50–53; *see id.* at 1:53–62, Figs. 2(a)–2(d). The sensitivities of the microphones may not match due to manufacturing differences or “drift over time.” *See id.* at 2:3–10, 2:35–37. “Hence, once matched microphones can become mismatched over time.” *Id.* at 2:10–11.

Hou purports to address that problem by providing “improved approaches to matching sensitivities of microphones in multi-microphone directional processing systems.” Ex. 1008, 2:44–46, 4:41–43, code (57); *see id.* at 2:38–40. Hou’s approaches “adaptively match microphone sensitivities” by adaptively measuring “a sensitivity difference between microphones” and then compensating (or correcting) “an electronic sound signal from one or more of the microphones.” *Id.* at 2:46–48, 4:43–44, 4:50–54, code (57). “As a result of the adaptive processing, the microphones ‘effectively’ become matched and remain matched over time and while in use.” *Id.* at 4:54–56; *see id.* at 2:48–50, 4:45–47, code (57). Hou’s approaches permit “multi-microphone directional processing systems to achieve superior directionality and consistent Signal-to-Noise Ratio

(SNR) across all conditions.” *Id.* at 4:57–60, 9:38–41. Accordingly, the “directional noise suppression is robust.” *Id.* at 2:48, 4:44–45, code (57); *see id.* at 9:29–38.

Hou’s Figure 5 (reproduced below) depicts a two-microphone directional processing system:

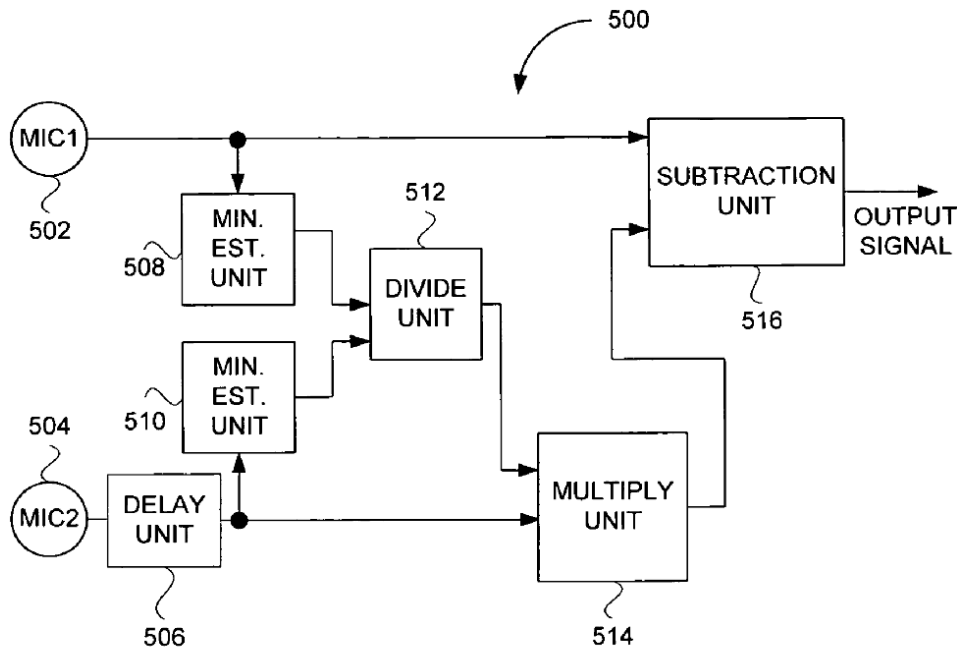


FIG. 5

Figure 5 illustrates two-microphone directional processing system 500 including the following components:

- (1) first microphone 502 that “produces a first electronic sound signal”;
- (2) second microphone 504 that “produces a second electronic sound signal”;
- (3) delay unit 506 that “delays the second electronic sound signal”;
- (4) first minimum estimate unit 508;

- (5) second minimum estimate unit 510;
- (6) divide unit 512;
- (7) multiplication unit 514; and
- (8) subtraction unit 516.

Ex. 1008, 4:17–19, 5:26–36, 5:43–45, 5:50–52, Fig. 5.

First minimum estimate unit 508 “estimates the minimum for the first electronic sound signal.” Ex. 1008, 5:36–38. Second minimum estimate unit 510 “estimates the minimum of the second electronic sound signal.” *Id.* at 5:38–39. “Typically, these minimums are measured over a time constant duration, such that the minimum is a relatively long-term minimum.” *Id.* at 5:39–41.

Divide unit 512 “produces a quotient by dividing the first minimum estimate by the second minimum estimate.” Ex. 1008, 5:42–43. The “quotient represents a scaling amount that is sent to” multiplication unit 514. *Id.* at 5:43–45. “The second electronic sound signal is then multiplied with the scaling amount to produce a compensated sound signal.” *Id.* at 5:45–47. “The compensated sound signal is thus compensated (or corrected) for the relative difference in sensitivity between the mismatched first and second microphones 502 and 504.” *Id.* at 5:47–50.

Subtraction unit 516 “subtracts the compensated electronic sound signal from the first electronic sound signal to produce an output signal.” Ex. 1008, 5:50–52, Fig. 5. Thus, “the output signal has been processed by the two-microphone directional processing system 500 to have robust directionality despite a mismatch between the first and second microphones 502 and 504.” *Id.* at 5:52–56.

3. INDEPENDENT CLAIM 29

(a) Preamble [29pre]

Claim 29 recites “[a] method comprising.” Ex. 1001, 38:33.

Petitioner contends that Avendano ’880 discloses claim 29’s preamble because Avendano ’880 discloses “an exemplary method for utilizing ILD of omni-direction microphones for noise suppression and speech enhancement.” Pet. 34 (quoting Ex. 1005, 9:43–45) (citing Ex. 1003 ¶ 127; Ex. 1005, 3:12–7:9, 9:43–10:25, Figs. 1a–1b, 2–3, 4a–4b, 5–7).

At this stage of the proceeding, Patent Owner makes no arguments specific to claim 29’s preamble. *See, e.g.*, Prelim. Resp. 5–10. Nonetheless, the burden remains on Petitioner to demonstrate unpatentability. *See Dynamic Drinkware, LLC v. Nat’l Graphics, Inc.*, 800 F.3d 1375, 1378 (Fed. Cir. 2015).

Generally, a preamble does not limit a claim. *Allen Eng’g Corp. v. Bartell Indus., Inc.*, 299 F.3d 1336, 1346 (Fed. Cir. 2002). We need not decide whether claim 29’s preamble limits the claim because Petitioner establishes sufficiently for purposes of institution that Avendano ’880 discloses claim 29’s preamble. *See* Pet. 34; Ex. 1003 ¶¶ 51–60, 127. Specifically, Avendano ’880 discloses methods “for utilizing inter-microphone level differences (ILD) to attenuate noise and enhance speech.” *See, e.g.*, Ex. 1005, 2:5–7, 3:13–17, 9:43–45, code (57); *see also* Ex. 1003 ¶¶ 51–60, 127.

(b) Limitation [29a]

Claim 29 recites “[29a] forming a first virtual microphone.” Ex. 1001, 38:34.

Petitioner contends that Avendano '880 discloses limitation [29a] because Avendano '880's audio device 104 forms a cardioid primary signal C_f corresponding to a "first virtual microphone" as a combination of the primary acoustic signal X_1 and the secondary acoustic signal X_2 . Pet. 34–35 (citing Ex. 1003 ¶¶ 128–129; Ex. 1005, 4:20–6:10, Figs. 4a–4b). To support its contention, Petitioner provides an annotated version of Avendano '880's Figure 4a as reproduced below (Pet. 35):

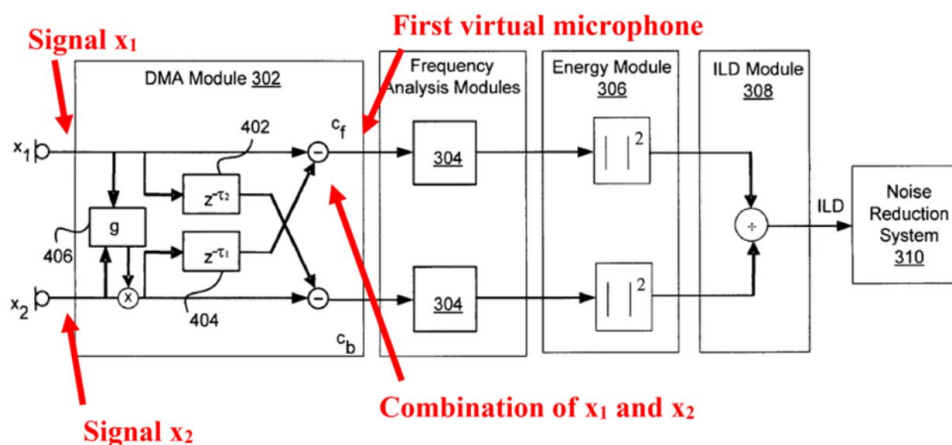


FIG. 4a

Figure 4a illustrates one implementation of audio processing engine 204 including various modules. Ex. 1005, 2:61–63, 5:15–17, 6:36–40, Fig. 4a. The above annotated version of Figure 4a includes the following annotations:

- (1) the red annotation "Signal x_1 " identifying the electric signal x_1 corresponding to the primary acoustic signal X_1 ;
- (2) the red annotation "Signal x_2 " identifying the electric signal x_2 corresponding to the secondary acoustic signal X_2 ;
- (3) the red annotation "Combination of x_1 and x_2 " identifying a component that combines the electric signal x_1 corresponding to the primary acoustic signal X_1 with a gain-adjusted and delayed version of the electric signal x_2 corresponding to the secondary acoustic signal X_2 ; and

- (4) the red annotation “First virtual microphone” identifying the cardioid primary signal C_f , i.e., at the output of the component that combines the electric signal x_1 corresponding to the primary acoustic signal X_1 with the gain-adjusted and delayed version of the electric signal x_2 corresponding to the secondary acoustic signal X_2 .

See Pet. 35; Ex. 1003 ¶ 129.

At this stage of the proceeding, Patent Owner makes no arguments specific to limitation [29a]. See, e.g., Prelim. Resp. 5–10. Nonetheless, the burden remains on Petitioner to demonstrate unpatentability. See *Dynamic Drinkware*, 800 F.3d at 1378.

Based on the current record and for the reasons advanced by Petitioner and supported by Dr. Kenny’s testimony, Petitioner establishes sufficiently for purposes of institution that the combined disclosures in Avendano ’880 and Hou teach limitation [29a]. See Pet. 34–35; Ex. 1003 ¶¶ 128–129.

(c) Limitation [29b]

Claim 29 recites as follows:

[29b] forming a filter by generating a first quantity by applying a calibration to a second signal of a second physical microphone, generating a second quantity by applying the delay to a first signal of a first physical microphone, and forming the filter as a ratio of the first quantity to the second quantity.

Ex. 1001, 38:35–40.

Petitioner contends that the combined disclosures in Avendano ’880 and Hou teach limitation [29b]. See Pet. 35–41. Specifically, Petitioner asserts that Hou’s processing system 500 would form a “filter” according to limitation [29b] based on Avendano ’880’s electric signal x_1 corresponding

to the primary acoustic signal X_1 and electric signal x_2 corresponding to the secondary acoustic signal X_2 by processing the signals x_1 and x_2 as follows:

- (1) generate a “first quantity” by either (i) applying Avendano ’880’s gain factor “g” or a multiplexer (or both) to the signal x_2 and then processing the calibrated signal x_2 with Hou’s first minimum estimate unit 508 to produce the “first quantity” or (ii) applying Hou’s first minimum estimate unit 508 directly to the signal x_2 to produce the “first quantity”;
- (2) generate a “second quantity” by applying Hou’s delay unit 506 to the signal x_1 and then processing the delayed signal x_1 with Hou’s second minimum estimate unit 510 to produce the “second quantity”; and
- (3) form the “filter” with Hou’s divide unit 512 that “produces a quotient by dividing” the first minimum estimate (the “first quantity” at the output of first minimum estimate unit 508) by the second minimum estimate (the “second quantity” at the output of second minimum estimate unit 510).

Id. at 36–41 (citing Ex. 1003 ¶¶ 130–137; Ex. 1005, 3:27–55, 4:20–27, 5:36–39, 9:54–59, Figs. 1a–1b, 4a–4b; Ex. 1008, 5:32–33, 5:36–38, 5:42–45, Fig. 5); *see id.* at 20–28.

To show how Hou’s processing system 500 would form a “filter” according to limitation [29b] by processing the electric signals x_1 and x_2 from Avendano ’880, Petitioner provides an annotated version of Hou’s Figure 5 as reproduced below (Pet. 41):

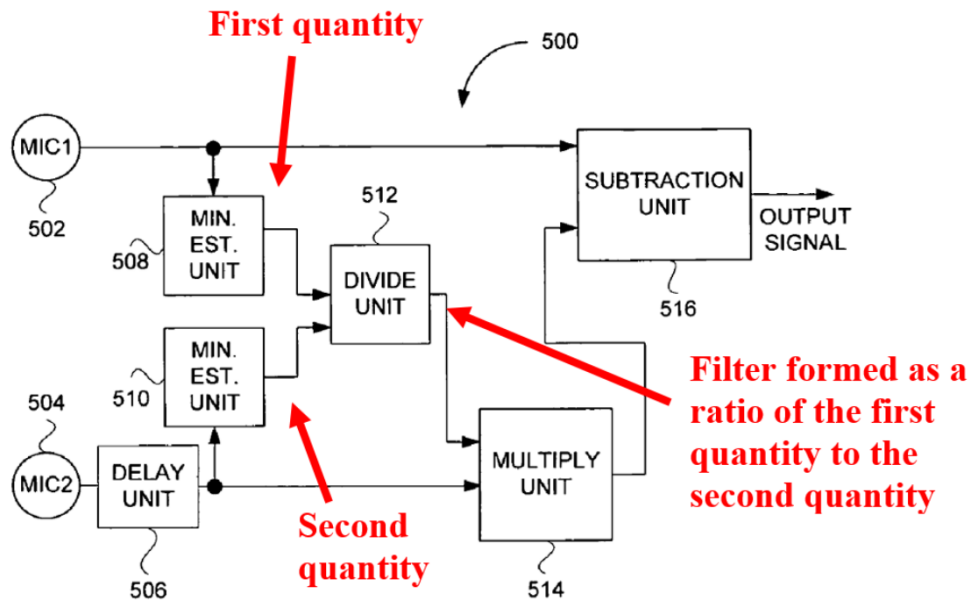
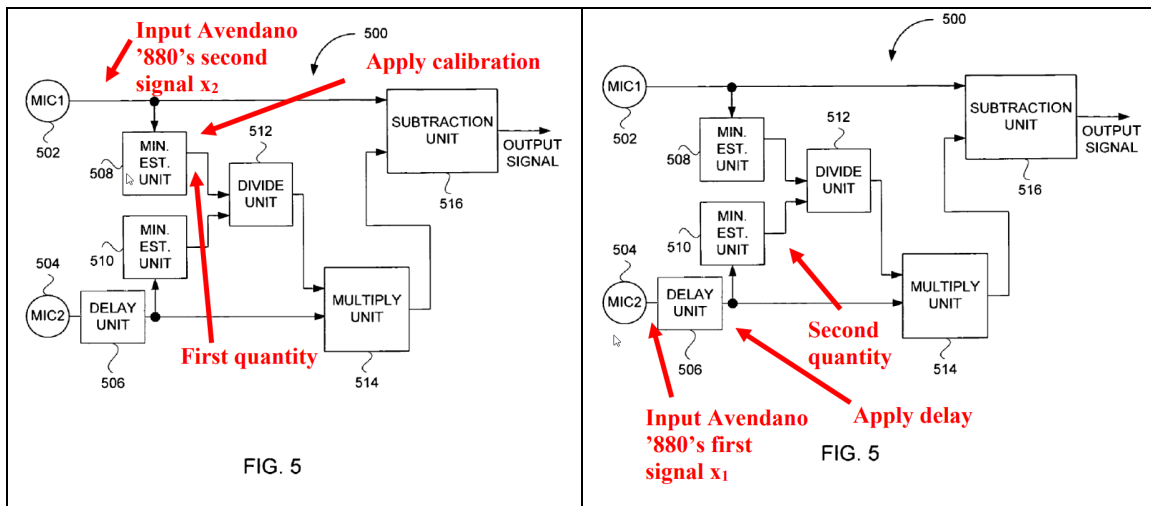


Figure 5 illustrates two-microphone directional processing system 500 including multiple components. Ex. 1008, 4:17–19, 5:26–36, 5:43–45, 5:50–52, Fig. 5. The above annotated version of Figure 5 includes the following annotations:

- (1) the red annotation “First quantity” identifying the output of first minimum estimate unit 508, which receives the signal x_2 as an input in the proposed combination;
- (2) the red annotation “Second quantity” identifying the output of second minimum estimate unit 510, which receives a delayed version of the signal x_1 as an input in the proposed combination; and
- (3) the red annotation “Filter formed as a ratio of the first quantity to the second quantity” identifying the output of divide unit 512.

See Pet. 41; Ex. 1003 ¶ 137.

Additionally, to show how Hou’s processing system 500 would generate a “first quantity” and a “second quantity” by processing the electric signals x_1 and x_2 from Avendano ’880, Dr. Kenny provides the annotated versions of Hou’s Figure 5 as reproduced below (Ex. 1003 ¶¶ 135–136):



The annotated version of Figure 5 above on the left illustrates generating a “first quantity” using the electric signal x_2 , and the annotated version of Figure 5 above on the right illustrates generating a “second quantity” using the electric signal x_1 . See Ex. 1003 ¶¶ 135–136; see also Pet. 39–40.

The annotated version of Figure 5 above on the left includes the following annotations:

- (1) the red annotation “Input Avendano '880's second signal x_2 ” identifying the input to first minimum estimate unit 508 (and the output of first microphone 502);
- (2) the red annotation “Apply calibration” identifying first minimum estimate unit 508; and
- (3) the red annotation “First quantity” identifying the output of first minimum estimate unit 508.

See Ex. 1003 ¶ 135; see also Pet. 39.

The annotated version of Figure 5 above on the right includes the following annotations:

- (1) the red annotation “Input Avendano '880's first signal x_1 ” identifying the input to delay unit 506 (and the output of second microphone 504);

- (2) the red annotation “Apply delay” identifying delay unit 506; and
- (3) the red annotation “Second quantity” identifying the output of second minimum estimate unit 510.

See Ex. 1003 ¶ 136; *see also* Pet. 40.

At this stage of the proceeding, Patent Owner makes no arguments specific to limitation [29b]. *See, e.g.*, Prelim. Resp. 5–10. Nonetheless, the burden remains on Petitioner to demonstrate unpatentability. *See Dynamic Drinkware*, 800 F.3d at 1378.

Based on the current record and for the reasons advanced by Petitioner and supported by Dr. Kenny’s testimony, Petitioner establishes sufficiently for purposes of institution that the combined disclosures in Avendano ’880 and Hou teach limitation [29b]. *See* Pet. 35–41; Ex. 1003 ¶¶ 130–137.

(d) Limitation [29c]

Claim 29 recites “[29c] forming a second virtual microphone by applying the filter to the first signal to generate a first intermediate signal, and summing the first intermediate signal and the second signal.” Ex. 1001, 38:41–44.

Petitioner contends that the combined disclosures in Avendano ’880 and Hou teach limitation [29c]. *See* Pet. 41–49. Additionally, Petitioner describes two example modifications to show how the combined disclosures do so. *Id.* at 41–47.

For the first example modification, Petitioner explains that:

- (1) Hou’s divide unit 512 forms a filter, e.g., as “a ratio of the first quantity to the second quantity”;

- (2) this filter is applied to Avendano '880's electric signal x_1 corresponding to the primary acoustic signal X_1 by Hou's multiplication unit 514 to generate a "first intermediate signal"; and
- (3) Hou's subtraction unit 516 subtracts "the first intermediate signal" from Avendano '880's electric signal x_2 corresponding to the secondary acoustic signal X_2 to form a "second virtual microphone."

Pet. 41–43 (citing Ex. 1003 ¶¶ 140–141; Ex. 1008, 5:42–56, Fig. 5); *see id.* at 23–25. Petitioner asserts that Hou's subtraction unit 516 would sum (i) Avendano '880's electric signal x_2 and (ii) the inverse of "the first intermediate signal" consistent with the '611 patent's disclosure that "varying the magnitude and sign of the delays and gains of the processing paths leads to a wide variety of virtual microphones." *Id.* at 43–44 (emphasis by Petitioner) (quoting Ex. 1001, 21:39–41).

Petitioner shows the first example modification with an annotated version of Hou's Figure 5 as reproduced below (Pet. 43):

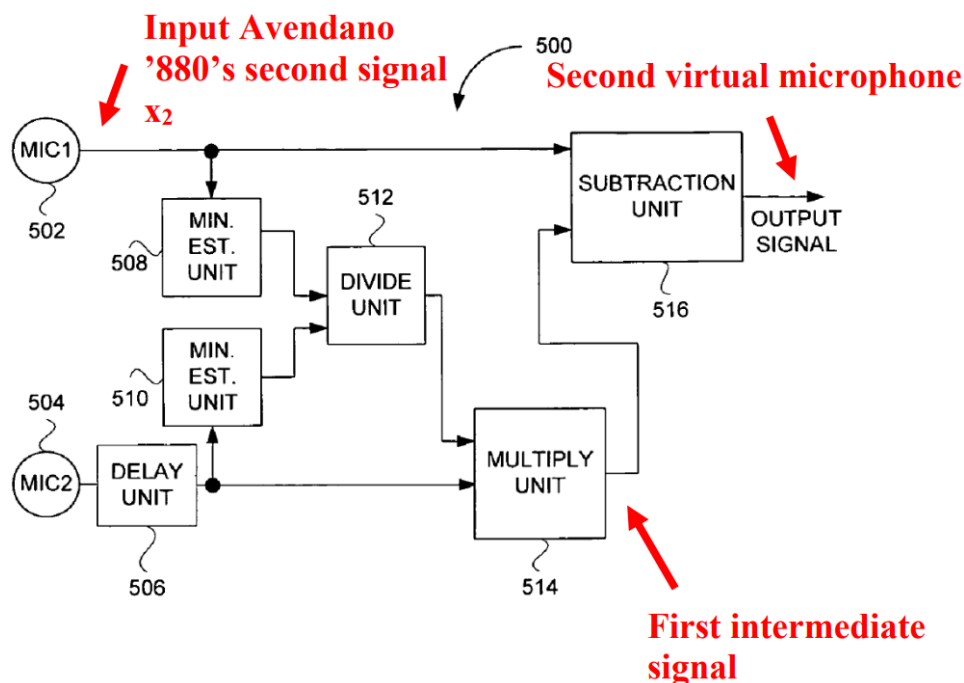


Figure 5 illustrates two-microphone directional processing system 500 including multiple components. Ex. 1008, 4:17–19, 5:26–36, 5:43–45, 5:50–52, Fig. 5. The above annotated version of Figure 5 includes the following annotations:

- (1) the red annotation “Input Avendano ’880’s second signal x_2 ” identifying the upper input to subtraction unit 516 (and the output of first microphone 502);
- (2) the red annotation “First intermediate signal” identifying the output of multiplication unit 514 and the lower input to subtraction unit 516; and
- (3) the red annotation “Second virtual microphone” identifying the output of subtraction unit 516.

See Pet. 43; Ex. 1003 ¶ 141.

For the second example modification, Petitioner explains that:

- (1) Hou’s divide unit 512 forms a filter, e.g., as “a ratio of the first quantity to the second quantity”;
- (2) this filter is applied to Avendano ’880’s electric signal x_1 corresponding to the primary acoustic signal X_1 after the signal x_1 passes through Avendano ’880’s delay node 402 to generate a “first intermediate signal”; and
- (3) the component in Avendano ’880 that combines signals to form the cardioid secondary signal C_b subtracts “the first intermediate signal” from Avendano ’880’s electric signal x_2 corresponding to the secondary acoustic signal X_2 to form a “second virtual microphone.”

Pet. 45–47 (citing Ex. 1003 ¶¶ 149–150; Ex. 1005, 5:15–35, Fig. 4a; Ex. 1008, 5:42–45, Fig. 5); *see id.* at 28–33. Petitioner asserts that the filter formed by Hou’s divide unit 512 would equalize the electric signal x_1 with the electric signal x_2 . *Id.* at 45 (citing Ex. 1003 ¶ 149).

Petitioner shows the second example modification with a modified and annotated version of Avendano '880's Figure 4a as reproduced below (Pet. 46):

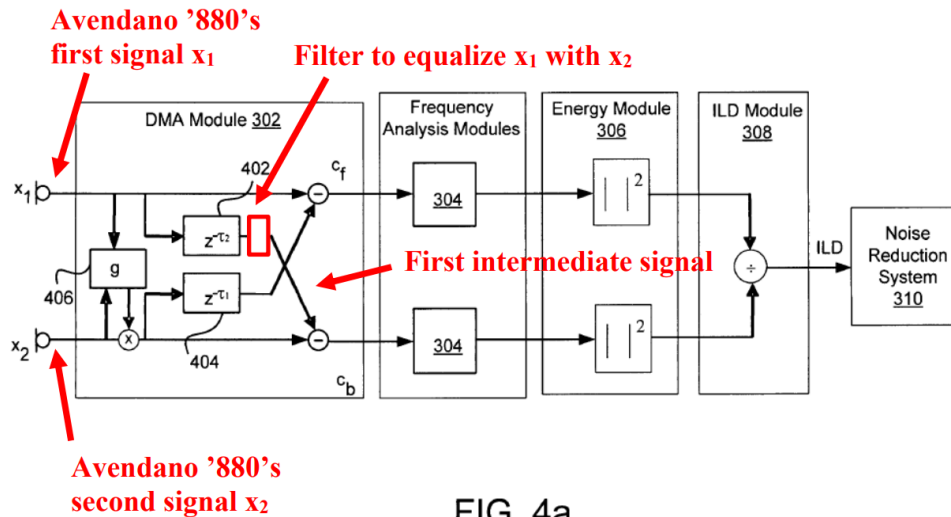


FIG. 4a

Figure 4a illustrates one implementation of audio processing engine 204 including various modules. Ex. 1005, 2:61–63, 5:15–17, 6:36–40, Fig. 4a. The above modified and annotated version of Figure 4a includes the following modification and annotations:

- (1) the red annotation "Avendano '880's first signal x_1 " identifying the electric signal x_1 corresponding to the primary acoustic signal X_1 ;
- (2) the red annotation "Avendano '880's second signal x_2 " identifying the electric signal x_2 corresponding to the secondary acoustic signal X_2 ;
- (3) a red box added at the output of delay node 402;
- (4) the red annotation "Filter to equalize x_1 with x_2 " identifying the added red box; and
- (5) the red annotation "First intermediate signal" identifying the output of the added red box (and an input to the component that combines signals to form the cardioid secondary signal C_b).

See Pet. 46; Ex. 1003 ¶ 149.

At this stage of the proceeding, Patent Owner makes no arguments specific to limitation [29c]. See, e.g., Prelim. Resp. 5–10. Nonetheless, the burden remains on Petitioner to demonstrate unpatentability. See *Drinkware*, 800 F.3d at 1378.

Based on the current record and for the reasons advanced by Petitioner and supported by Dr. Kenny’s testimony, Petitioner establishes sufficiently for purposes of institution that the combined disclosures in Avendano ’880 and Hou teach limitation [29c]. See Pet. 41–49; Ex. 1003 ¶¶ 138–155.

(e) Limitation [29d]

Claim 29 recites “[29d] generating a ratio of energies of the first virtual microphone and the second virtual microphone and detecting acoustic voice activity using the ratio.” Ex. 1001, 38:45–47.

Petitioner contends that Avendano ’880 (and in combination with Hou) discloses limitation [29d]. See Pet. 49–51. Specifically, Petitioner asserts that Avendano ’880’s audio device 104 determines the following:

- (1) the energy of the first virtual microphone E_f “by integrating the square of the first virtual microphone over time” using the equation $E_f(t, \omega) = \int_{\text{frame}} |C_f(t', \omega)|^2 dt'$;
- (2) the energy of the second virtual microphone E_b “by integrating the square of the second virtual microphone over time” using the equation $E_b(t, \omega) = \int_{\text{frame}} |C_b(t', \omega)|^2 dt'$; and
- (3) the energy ratio or Inter-Level Difference (ILD) “by dividing the energy of the first virtual microphone by the energy of the second virtual microphone.”

Pet. 49–50 (citing Ex. 1003 ¶¶ 158–160; Ex. 1005, 5:49–6:5, 6:8–34).

Petitioner also asserts that Avendano '880 “compares the energy ratio (ILD) to a ‘threshold’ to detect acoustic voice activity,” e.g., “the presence or absence of speech.” Pet. 50 (citing Ex. 1003 ¶¶ 161–164; Ex. 1005, 6:35–7:9). According to Petitioner, an energy ratio below the threshold results in setting the value λ_1 to zero “indicating an absence of speech,” while an energy ratio above the threshold results in setting the value λ_1 to one “indicating a presence of speech.” *Id.* at 50–51 (citing Ex. 1003 ¶ 164; Ex. 1005, 6:58–7:3).

At this stage of the proceeding, Patent Owner makes no arguments specific to limitation [29d]. *See, e.g.*, Prelim. Resp. 5–10. Nonetheless, the burden remains on Petitioner to demonstrate unpatentability. *See Dynamic Drinkware*, 800 F.3d at 1378.

Based on the current record and for the reasons advanced by Petitioner and supported by Dr. Kenny’s testimony, Petitioner establishes sufficiently for purposes of institution that Avendano '880 (and in combination with Hou) discloses limitation [29d]. *See* Pet. 49–51; Ex. 1003 ¶¶ 156–164.

(f) Alleged Reasons for Combining the Teachings of the References

Petitioner identifies reasons that would have prompted an ordinarily skilled artisan to combine Hou’s teachings with Avendano '880’s teachings in the way Petitioner proposes. *See* Pet. 19–33, 41–49. Additionally, Petitioner explains that an ordinarily skilled artisan “would have had a reasonable expectation of success” in combining the teachings of the references. *Id.* at 33. Dr. Kenny’s testimony supports Petitioner’s positions. *See* Ex. 1003 ¶¶ 86–126, 139–155.

For example, Dr. Kenny explains that Avendano '880's system includes DMA module 302 that:

- (1) applies a gain factor "g" to the electric signal x_2 "to equalize the signal levels" of the electric signals x_1 and x_2 and advantageously avoid "loss of performance when the microphone signals have different levels"; and
- (2) "uses the equalized signal to generate directional signals," such as (i) a cardioid primary signal C_f having a directional pattern for receiving sounds "within a front cardioid region around the audio device 104" and (ii) a cardioid secondary signal C_b signal having a directional pattern for receiving sounds "within a back cardioid region around the audio device 104."

Ex. 1003 ¶¶ 91–92 (quoting Ex. 1005, 5:36–39) (citing Ex. 1005, 4:41–52, 5:25–39, 9:54–59, Figs. 4a–4b, 6).

Dr. Kenny also explains that "Hou describes enhancing speech and attenuating noise using similar techniques as those described by Avendano '880" and that Hou's system 500 serves a "similar function" as Avendano '880's DMA module 302. Ex. 1003 ¶ 93. He identifies that "similar function" as "compensat[ing] (or correct[ing]) for the relative difference in sensitivity between . . . mismatched first and second microphones" because (1) "the directionality is substantially degraded" and (2) "the ability to suppress interference coming from a particular direction is poor" when the microphone sensitivities "are not properly matched." *Id.* (alterations by Dr. Kenny) (quoting Ex. 1008, 1:50–53, 5:48–50). Dr. Kenny additionally explains that "like Avendano '880, Hou processes the compensated or corrected signal to generate a directional signal (e.g., 'an output signal . . . hav[ing] robust directionality') to aid in 'directional noise suppression.'" *Id.*

¶ 94 (alterations by Dr. Kenny) (emphasis omitted) (quoting Ex. 1008, 2:48, 4:44–45, 5:53–55).

Dr. Kenny testifies that an ordinarily skilled artisan “would have recognized” that (1) “Avendano ’880 teaches the importance of equalizing signal levels of signals generated by two respective microphones” and (2) Hou discloses system 500 “for performing precisely this function.” Ex. 1003 ¶¶ 95, 179. According to Dr. Kenny, an ordinarily skilled artisan “would have been motivated to implement” aspects of Hou’s system 500 (containing “signal compensation or correction components”) in Avendano ’880’s system to “improve the equalization of the signal output by the microphones (e.g., to generate a signal ‘hav[ing] robust directionality’ and to enhance ‘directional noise suppression’).” *Id.* ¶ 95 (alteration by Dr. Kenny) (emphasis omitted) (quoting Ex. 1008, 2:48, 4:44–45, 5:54–55).

Dr. Kenny testifies that an ordinarily skilled artisan “would have found it obvious to implement” aspects of Hou’s system 500 along with Avendano ’880’s gain factor “g” because the skilled artisan “would have recognized” that the system 500 and the gain factor “g” are “configured to equalize the signal levels of microphone signals at different respective stages in a signal processing operation.” Ex. 1003 ¶ 97 (citing Ex. 1005, 5:36–39, 9:54–59, Figs. 4a–4b; Ex. 1008, 5:17–6:22, Fig. 5); *see id.* ¶¶ 110, 148, 155. Regarding the signal-processing stage for the gain factor “g,” Dr. Kenny explains that the gain factor “g” is:

- (1) “configured to perform signal equalization at an initial stage of a signal processing operation, in particular prior to performing any other signal processing tasks”; and
- (2) suitable “to account for a general ‘loss of performance when the microphone signals have different levels.’”

Id. ¶ 98 (quoting Ex. 1005, 5:37–39) (citing Ex. 1005, 5:36–39, 9:54–59, Figs. 4a–4b). Regarding the signal-processing stage for the system 500, Dr. Kenny explains that the system 500 is:

- (1) “configured to perform a signal equalization at a late stage of a signal processing operation,” e.g., “just prior to combining two signals together to create a combined signal having ‘robust directionality’”; and
- (2) suitable “for performing an additional signal compensation or correction for two microphone signals, such that the directionality of a combination of the two signals is enhanced.”

Id. ¶ 99 (quoting Ex. 1008, 5:55) (citing Ex. 1008, 5:17–6:22, Fig. 5).

Dr. Kenny testifies that implementing aspects of Hou’s system 500 along with Avendano ’880’s gain factor “g” would have provided “benefits that may not be obtained with either of the two alone” because the system 500 and the gain factor “g” are “configured to equalize the signal levels of microphone signals at different respective stages in a signal processing operation.” Ex. 1003 ¶¶ 97, 110, 148, 155. He identifies those benefits as follows:

- (1) accounting for “a general ‘loss of performance when the microphone signals have different levels’ at an initial stage of the signal processing operation”;
- (2) permitting “a finer-tuning of the gain prior to the combination of the signals”; and
- (3) enhancing “the directionality of the combined microphone signals at the late stage of the signal processing operation.”

Id. ¶¶ 110, 148, 155 (quoting Ex. 1005, 5:37–39) (citing Ex. 1005, 5:36–39; Ex. 1008, 5:17–6:22, Fig. 5). According to Dr. Kenny, the combination would have “enhance[d] the signal equalization functionality of Avendano

'880's system," e.g., by allowing the system to "generate a signal having 'robust directionality' in a backwards direction." *Id.* ¶¶ 101, 111 (quoting Ex. 1008, 5:55).

Dr. Kenny describes two example modifications to show how an ordinarily skilled artisan would have combined Hou's teachings with Avendano '880's teachings to "enhance the signal equalization functionality of Avendano '880's system." Ex. 1003 ¶¶ 100–121; *see id.* ¶¶ 139–155. The first example modification parallels the first example modification discussed above for limitation [29c]. *Id.* ¶¶ 101–110, 140–148. The second example modification parallels the second example modification discussed above for limitation [29c] except that a filter according to Hou's teachings is added in two locations instead of just one location. *Id.* ¶¶ 111–121, 149–155. Dr. Kenny identifies the two locations and associated filter functions as follows: first, at the output of Avendano '880's delay node 402 to equalize the electric signal x_1 with the electric signal x_2 (as discussed above for limitation [29c]); and second, at the output of Avendano '880's delay node 404 to equalize the electric signal x_2 with the electric signal x_1 . *Id.* ¶¶ 112–113.

Dr. Kenny shows the second example modification with a modified and annotated version of Avendano '880's Figure 4a as reproduced below (Ex. 1003 ¶ 113):

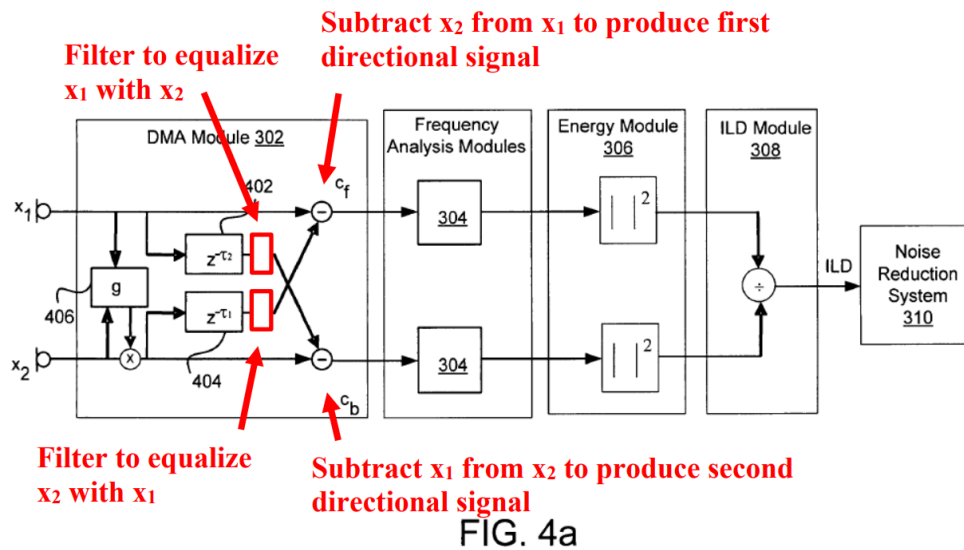


Figure 4a illustrates one implementation of audio processing engine 204 including various modules. Ex. 1005, 2:61–63, 5:15–17, 6:36–40, Fig. 4a. The above modified and annotated version of Figure 4a includes the following modifications and annotations:

- (1) a red box added at the output of delay node 402;
- (2) the red annotation “Filter to equalize x_1 with x_2 ” identifying the red box added at the output of delay node 402;
- (3) the red annotation “Subtract x_2 from x_1 to produce first directional signal” identifying the component that combines signals to form the cardioid primary signal C_f ;
- (4) a red box added at the output of delay node 404;
- (5) the red annotation “Filter to equalize x_2 with x_1 ” identifying the red box added at the output of delay node 404; and
- (6) the red annotation “Subtract x_1 from x_2 to produce second directional signal” identifying the component that combines signals to form the cardioid secondary signal C_b .

See Ex. 1003 ¶ 113.

Patent Owner disputes that an ordinarily skilled artisan would have been motivated to combine Hou's teachings with Avendano '880's teachings in the way Petitioner proposes. *See* Prelim. Resp. 5–10. Specifically, Patent Owner asserts that (1) “Hou is directed to the same purpose” as Avendano '880's gain factor “g,” i.e., “addressing the problem of mismatched microphone sensitivities,” and (2) each reference “calculates and applies a gain.” *Id.* at 6–7. According to Patent Owner, the record is “entirely bereft of any motivation to use two gain adjustments.” *Id.* at 7. Patent Owner also asserts that nothing indicates that “Hou should be operated in conjunction with another gain compensation system like that of Avendano.” *Id.* at 9. Regarding the “benefits that may not be obtained with either of the two alone,” Patent Owner asserts that the portions of Avendano '880 and Hou that Petitioner and Dr. Kenny cite “do not say anything of the sort.” *Id.*

Patent Owner further asserts that there is “no evidence that Avendano is only useful at an ‘initial’ stage or that Hou is designed for use at a ‘late stage.’” Prelim. Resp. 7–8 (citing Ex. 1003 ¶ 98; Ex. 1005, 5:36–37, 9:54–59; Ex. 1008, 5:17–22, 6:16–18, Fig. 5).

Additionally, Patent Owner contends that the “proposed combination would apply Hou's gain immediately after Avendano's gain.” Prelim. Resp. 7–8.

Based on the current record, Petitioner establishes sufficiently for purposes of institution that an ordinarily skilled artisan would have had reasons to combine Hou's teachings with Avendano '880's teachings in the way Petitioner proposes. *See* Pet. 19–33; Ex. 1003 ¶¶ 86–121, 139–155. As Dr. Kenny testifies, an ordinarily skilled artisan “would have been motivated to implement” aspects of Hou's system 500 (containing “signal

compensation or correction components”) in Avendano ’880’s system to “improve the equalization of the signal output by the microphones.” Ex. 1003 ¶ 95. As Dr. Kenny also testifies, the combination would have “enhance[d] the signal equalization functionality of Avendano ’880’s system,” e.g., by allowing the system to “generate a signal having ‘robust directionality’ in a backwards direction.” *Id.* ¶¶ 101, 111; *see id.* ¶¶ 95, 110, 148, 155.

Any “need or problem known in the field of endeavor at the time of invention and addressed by the patent can provide a reason for combining” references. *KSR*, 550 U.S. at 420. At this stage of the proceeding, we are persuaded that “improv[ing] the equalization of the signal output by the microphones” by “enhanc[ing] the signal equalization functionality of Avendano ’880’s system” suffices as a reason for combining Hou’s teachings with Avendano ’880’s teachings. *See* Ex. 1003 ¶¶ 95, 101, 110–111, 148, 155. The “desire to enhance commercial opportunities by improving a product or process is universal” *DyStar Textilfarben GmbH v. C.H. Patrick Co.*, 464 F.3d 1356, 1368 (Fed. Cir. 2006).

We disagree with Patent Owner that the record is “entirely bereft of any motivation to use two gain adjustments.” *See* Prelim. Resp. 9. Dr. Kenny testifies that an ordinarily skilled artisan “would have recognized” that Hou’s system 500 and Avendano ’880’s gain factor “g” are “configured to equalize the signal levels of microphone signals at different respective stages in a signal processing operation.” Ex. 1003 ¶¶ 97, 110, 148, 155 (citing Ex. 1005, 5:36–39, 9:54–59, Figs. 4a–4b; Ex. 1008, 5:17–6:22, Fig. 5). In particular, he explains that:

- (1) the gain factor “g” is “configured to perform signal equalization at an initial stage of a signal processing operation, in particular prior to performing any other signal processing tasks”; and
- (2) the system 500 is “configured to perform a signal equalization at a late stage of a signal processing operation,” e.g., “just prior to combining two signals together to create a combined signal having ‘robust directionality.’”

Id. ¶¶ 98–99 (citing Ex. 1005, 5:36–39, 9:54–59, Figs. 4a–4b; Ex. 1008, 5:17–6:22, Fig. 5).

We disagree with Patent Owner that there is “no evidence that Avendano is only useful at an ‘initial’ stage or that Hou is designed for use at a ‘late stage.’” *See* Prelim. Resp. 7–8. Dr. Kenny cites the references themselves to support his testimony that Hou’s system 500 and Avendano ’880’s gain factor “g” are “configured to equalize the signal levels of microphone signals at different respective stages in a signal processing operation.” *See* Ex. 1003 ¶¶ 97–99, 110, 148, 155 (citing Ex. 1005, 5:36–39, 9:54–59, Figs. 4a–4b; Ex. 1008, 5:17–6:22, Fig. 5).

As this stage, the record favors Petitioner’s position. Consistent with Dr. Kenny’s testimony, Avendano ’880 discloses that gain module 406 in Figure 4a applies the gain factor “g” to the electric signal x_2 corresponding to the secondary acoustic signal X_2 at an initial signal-processing stage, i.e., before further processing by DMA module 302 and any downstream module. Ex. 1005, 5:36–39, 8:52–60, 9:16–22, Fig. 4a; *see* Ex. 1003 ¶¶ 97–98, 110, 148, 155. Also consistent with Dr. Kenny’s testimony, Hou discloses that system 500 compensates (or corrects) a “second electronic sound signal” by a “scaling amount” at a later signal-processing stage, i.e.,

just before subtraction unit 516 subtracts the “compensated electronic sound signal” from a “first electronic sound signal” to produce an “output signal.” Ex. 1008, 5:26–52, Fig. 5; *see* Ex. 1003 ¶¶ 97, 99, 110, 148, 155. This issue is better resolved at trial on a more developed record.

Additionally, based on the current record, we disagree with Patent Owner that the “proposed combination would apply Hou’s gain immediately after Avendano’s gain.” *See* Prelim. Resp. 7–8. In the first example modification, Avendano ’880’s gain module 406 would apply the gain factor “g” to the electric signal x_2 corresponding to the secondary acoustic signal X_2 , while Hou’s multiplication unit 514 would apply a filter formed by Hou’s divide unit 512 to the electric signal x_1 corresponding to the primary acoustic signal X_1 . *See* Pet. 23–28, 42–45; Ex. 1003 ¶¶ 101–110, 140–148. Hence, in that proposed combination, different gains would be applied to different signals.

In the second example modification, Avendano ’880’s gain module 406 would apply the gain factor “g” to the electric signal x_2 corresponding to the secondary acoustic signal X_2 , while a filter formed by Hou’s divide unit 512 would be applied to the electric signal x_1 corresponding to the primary acoustic signal X_1 after the electric signal x_1 passes through Avendano ’880’s delay node 402. *See* Pet. 28–33, 45–49; Ex. 1003 ¶¶ 111–121, 149–155. Also, a filter formed by Hou’s divide unit 512 would be applied to the gain-adjusted electric signal x_2 after the gain-adjusted electric signal x_2 passes through Avendano ’880’s delay node 404. *See* Pet. 28–31; Ex. 1003 ¶¶ 112–118. Hence, in that proposed combination, different gains would be applied to different signals.

(g) Preliminary Conclusion for Obviousness/Nonobviousness of Claim 29

As discussed above, Petitioner's analysis addresses how the combined disclosures in Avendano '880 and Hou teach claim 29's subject matter. *See supra* §§ IV.D.3(a)–(e). Additionally, Petitioner provides a reason with rational underpinning as to why an ordinarily skilled artisan would have been motivated to combine Hou's teachings with Avendano '880's teachings in the way Petitioner proposes and would have had a reasonable expectation of success. *See supra* § IV.D.3(f). Hence, Petitioner demonstrates a reasonable likelihood of prevailing in proving claim 29 unpatentable under § 103(a) as obvious over Avendano '880 and Hou.

4. DEPENDENT CLAIMS 30, 31, 33–37, AND 41–43

Claims 30, 31, 33–37, and 41–43 depend directly or indirectly from claim 29. Ex. 1001, 38:48–56, 38:62–39:8, 39:20–40:2.

For claims 30, 31, 33–37, and 41–43, Petitioner provides an analysis supported by Dr. Kenny's testimony about how the combined disclosures in Avendano '880 and Hou teach the claimed subject matter. *See* Pet. 51–69; Ex. 1003 ¶¶ 165–214.

At this stage of the proceeding, Patent Owner makes no arguments specific to claims 30, 31, 33–37, and 41–43. *See, e.g.,* Prelim. Resp. 5–10. Nonetheless, the burden remains on Petitioner to demonstrate unpatentability. *See Dynamic Drinkware*, 800 F.3d at 1378.

Based on the current record and for the reasons advanced by Petitioner and supported by Dr. Kenny's testimony, Petitioner establishes sufficiently for purposes of institution that the combined disclosures in Avendano '880 and Hou teach the subject matter in claims 30, 31, 33–37, and 41–43. *See* Pet. 51–69; Ex. 1003 ¶¶ 165–214. Hence, Petitioner

demonstrates a reasonable likelihood of prevailing in proving claims 30, 31, 33–37, and 41–43 unpatentable under § 103(a) as obvious over Avendano '880 and Hou.

5. INDEPENDENT CLAIM 44

Petitioner contends that the combined disclosures in Avendano '880 and Hou teach claim 44's subject matter for the same reasons that Petitioner contends that the combined disclosures teach claim 29's subject matter. *See* Pet. 69–70; *see also id.* at 34–51.

Patent Owner disputes that the combined disclosures in Avendano '880 and Hou teach claim 44's subject matter for the same reasons that Patent Owner disputes that the combined disclosures teach claim 29's subject matter. *See* Prelim. Resp. 5–10.

Based on the current record and for (1) the reasons advanced by Petitioner and supported by Dr. Kenny's testimony and (2) the reasons discussed above for claim 29, Petitioner establishes sufficiently for purposes of institution that the combined disclosures in Avendano '880 and Hou teach claim 44's subject matter. *See* Pet. 34–51, 69–70; Ex. 1003 ¶¶ 127–174, 215–219; *supra* §§ IV.D.3(a)–(f). Hence, Petitioner demonstrates a reasonable likelihood of prevailing in proving claim 44 unpatentable under § 103(a) as obvious over Avendano '880 and Hou.

E. Alleged Obviousness over Avendano '880, Avendano '252, and Hou: Claims 32 and 38–40

Petitioner contends that claims 32 and 38–40 are unpatentable under § 103(a) as obvious over Avendano '880, Avendano '252, and Hou. *See* Pet. 4, 70–91. Patent Owner disputes Petitioner's contentions. *See* Prelim. Resp. 10. Above, we provided overviews of Avendano '880 and Hou. *See*

supra §§ IV.D.1–IV.D.2. Below, we provide an overview of Avendano ’252, and then we consider the obviousness issues. As explained below, Petitioner establishes sufficiently for purposes of institution that the combined disclosures in Avendano ’880, Avendano ’252, and Hou teach the subject matter in claims 32 and 38–40.

1. OVERVIEW OF AVENDANO ’252 (EXHIBIT 1006)

Avendano ’252 is a U.S. patent titled “System and Method for Providing Close Microphone Adaptive Array Processing,” filed on March 31, 2008, and issued on June 19, 2012. Ex. 1006, codes (12), (22), (45), (54). The application for Avendano ’252 is a continuation-in-part of the application for Avendano ’880. *Compare* Ex. 1005, codes (21)–(22), *with* Ex. 1006, 1:8–9, code (63). Avendano ’252 incorporates Avendano ’880 by reference. Ex. 1006, 1:8–23. Avendano ’252 states that the invention “relates generally to audio processing and more particularly to adaptive array processing in close microphone systems.” *Id.* at 1:28–30; *see id.* at code (57). The close microphones may “comprise onmi-directional microphones.” *Id.* at 2:65–66.

Avendano ’252 explains that array processing (AP) technology “relies on accurate phase and/or level match of the microphones to create the desired cardioid patterns.” Ex. 1006, 3:8–10. Without proper calibration, however, “even a small phase mismatch between the microphones may cause serious deterioration of an intended directivity patterns [sic] which may in turn introduce distortion to an inter-microphone level difference (ILD) map and either produce speech loss or noise leakage at a system output.” *Id.* at 3:10–15. According to Avendano ’252, “[c]alibration for phase mismatch is essential for current AP technology to work given

observed mismatches in microphone responses inherent in the manufacturing processes.” *Id.* at 3:15–18.

Avendano ’252 explains that the invention generates simulated “directional patterns (i.e., cardioid patterns)” by processing signals received from close microphones and adapts the cardioid patterns to “compensate for microphone mismatch.” Ex. 1006, 2:66–3:3. The “adaptation may result in a null of a backward-facing cardioid pattern to be directed towards a desired audio source.” *Id.* at 3:3–5. “The resulting signals from the adaptation may then be utilized in a noise suppression system and/or speech enhancement system.” *Id.* at 3:5–7.

Avendano ’252’s Figure 3 (reproduced below) depicts an embodiment of an audio processing engine:

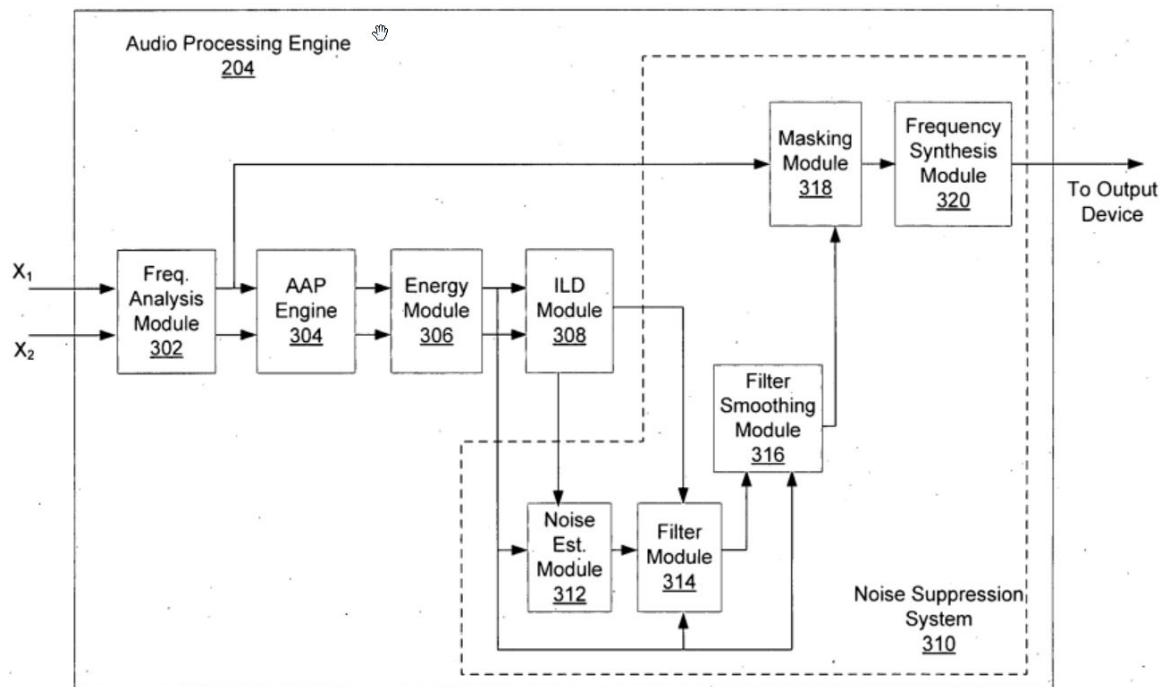


FIG. 3

Figure 3 illustrates one implementation of audio processing engine 204 including frequency-analysis module 302, adaptive array-processing (AAP)

engine 304, energy module 306, ILD module 308, and noise-suppression system 310. Ex. 1006, 2:47–48, 4:59–65, 5:18–20, 6:58–63, 7:26–28, 7:65–8:3, Fig. 3; *see id.* at 1:56. As Figure 3 shows, frequency-analysis module 302 receives an electric signal x_1 corresponding to a primary acoustic signal produced by primary microphone 106 (not shown) and an electric signal x_2 corresponding to a secondary acoustic signal produced by secondary microphone 108 (not shown). *Id.* at 3:33–36, 4:49–54, 4:62–65, Figs. 1a–1b, 3.

After receiving the electric signals x_1 and x_2 , frequency-analysis module 302 “mimics the frequency analysis of the cochlea (i.e., cochlear domain),” e.g., by separating the signals into frequency sub-bands. Ex. 1006, 4:65–5:3. To simulate the cochlear domain, frequency-analysis module 302 may employ a filter bank or similar mechanism, e.g., a short-time Fourier transform (STFT), sub-band filter banks, or modulated complex lapped transforms. *Id.* at 4:65–5:1, 5:6–10. After determining the sub-band signals, frequency-analysis module 302 forwards the sub-band signals to AAP engine 304. *Id.* at 5:18–20, Fig. 3.

AAP engine 304 adaptively processes “the primary and secondary signals to create synthetic directional patterns (i.e., synthetic directional microphone responses) for the close microphone array (e.g., primary and secondary microphones 106 and 108).” Ex. 1006, 5:20–24. “The directional patterns may comprise a forward-facing cardioid pattern based on the primary acoustic (sub-band) signal and a backward-facing cardioid pattern based on the secondary (sub-band) acoustic signal.” *Id.* at 5:24–28. The “sub-band signals may be adapted such that a null of the backward-facing cardioid pattern is directed towards the audio source.” *Id.* at 5:28–31.

“In essence, this processing replaces two cardioid or directional microphones with two omni-directional microphones.” *Id.* at 5:33–35.

Next, energy module 306 “computes energy (power) estimates or spectra associated with the cardioid signals.” Ex. 1006, 6:58–63. “The power estimates are computed based on a cardioid primary signal (c_1) of the forward-facing cardioid and cardioid secondary signal (c_2) of a backward facing cardioid during an interval of time for each frequency band.” *Id.* at 6:67–7:4. Each “power estimate may be mathematically determined by squaring and integrating an absolute value of the frequency analyzed cardioid signals.” *Id.* at 7:5–8.

Energy module 306 supplies the calculated energy (power) levels to ILD module 308. Ex. 1006, 7:26–28, Fig. 3. Then, ILD module 308 determines the ILD “in a non-linear manner by taking a ratio of the energy levels” according to the following equation where E_1 represents the “energy level associated with the primary microphone signal,” E_2 represents the “energy level associated with the secondary microphone signal,” “ n ” represents a time index, and “ k ” represents a frequency index:

$$ILD(n,k) = E_1(n,k)/E_2(n,k)$$

Id. at 7:8–25, 7:28–33. “Once the ILD is determined, the cardioid sub-band signals are processed through a noise suppression system 310.” *Id.* at 7:65–66, Fig. 3.

Avendano '252's Figure 4b (reproduced below) depicts an embodiment of an array-processing module:

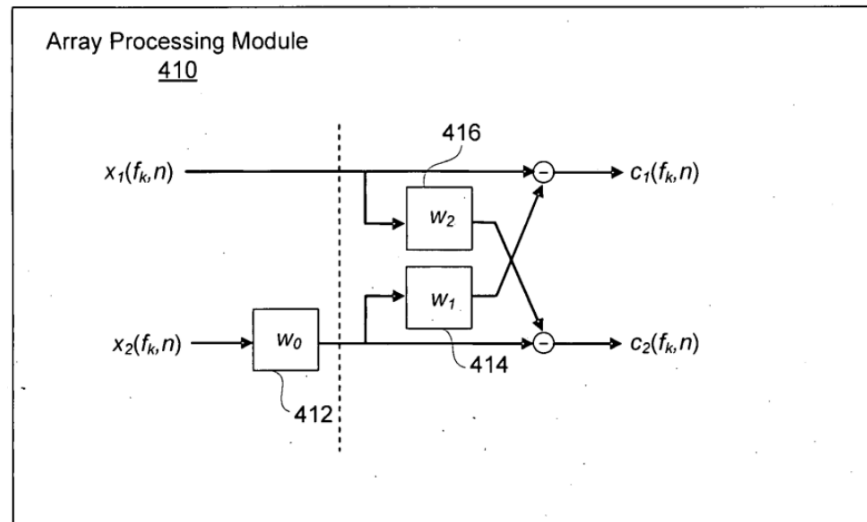


FIG. 4b

Figure 4b illustrates array-processing module 410 including equalization module 412, delay node 414, and delay node 416. Ex. 1006, 6:9–10, 6:33–34, 6:47–50, Fig. 4b.

Array-processing module 410 implements “two cardioid patterns, one facing front (i.e., forward-facing cardioid pattern) and one facing back (i.e., backward-facing cardioid pattern).” Ex. 1006, 6:12–15. Array-processing module 410 receives an electric signal $x_1(f_k, n)$ corresponding to a primary acoustic signal produced by primary microphone 106 and an electric signal $x_2(f_k, n)$ corresponding to a secondary acoustic signal produced by secondary microphone 108. *Id.* at 6:15–30, Fig. 4b.

Equalization module 412 applies an equalization coefficient w_0 to the signal $x_2(f_k, n)$. Ex. 1006, 6:30–34, Fig. 4b. The equalization coefficient w_0 “comprises a phase shift or time delay that aligns the two microphones 106

and 108 by modeling their phase mismatch.” *Id.* at 6:30–33. Additionally, “to create the two cardioid patterns,” delay node 414 applies a delay coefficient w_1 to the signal $x_2(f_k, n)$, and delay node 416 applies a delay coefficient w_2 to the signal $x_1(f_k, n)$. *Id.* at 6:39–41, 6:47–50, Fig. 4b. The delay coefficients w_1 and w_2 may “include the phase shifts due to acoustical properties of the audio device” and “exclude particular microphone properties.” *Id.* at 10:18–20; *see id.* at 6:44–47.

Avendano ’252 explains that “[p]attern generation using differential arrays (DA) requires use of fractional delays whose value may depend on a distance between the microphones.” Ex. 1006, 5:36–38. Hence, a delay node or delay line may introduce “a delay equal to an acoustic delay between the primary and secondary microphones 106 and 108.” *Id.* at 5:51–54.

Avendano '252's Figure 5 (reproduced below) depicts an embodiment of an AAP engine:

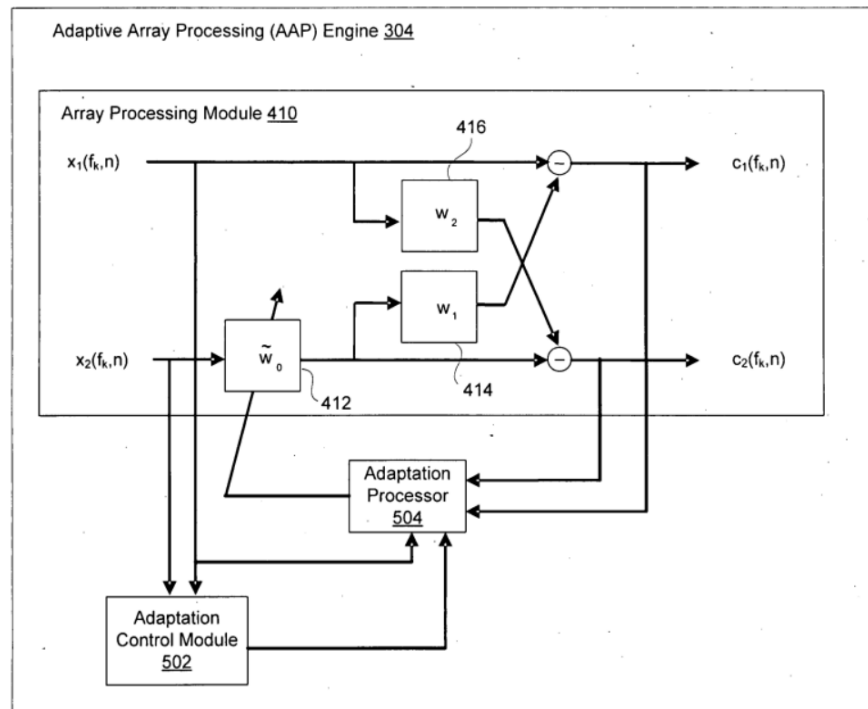


FIG. 5

Figure 5 illustrates AAP engine 304 including array-processing module 410, adaption-control module 502, and adaption processor 504. Ex. 1006, 2:52–53, 10:1–6, Fig. 5.

Like array-processing module 410 in Figure 4b, array-processing module 410 in Figure 5 includes equalization module 412, delay node 414, and delay node 416. Ex. 1006, 10:1–6, 10:53–59, Fig. 5. But equalization module 412 in Figure 5 applies an “adaptive equalization coefficient” to the signal $x_2(f_k, n)$ as determined by adaptation-control module 502 and adaptation processor 504. *Id.* at 10:3–6, Fig. 5. The “adaptive equalization coefficient” compensates for “microphone mismatch post-manufacturing.” *Id.* at 10:6–8; *see id.* at 2:22–23, 3:2–3.

Adaptation-control module 502 operates as a switch to activate adaptation processor 504. Ex. 1006, 10:9–11. The “adaptation may be triggered by identifying frames dominated by speech.” *Id.* at 10:12–13. For example, the “power ratio between the front-facing and back-facing cardioid signals produced by this array may be tracked and used to determine if a signal is active in the forward direction, in which case the adaptive equalization coefficient can be updated.” *Id.* at 10:20–24.

Adaptation-control module 502 may provide “a value of one” to permit adaptation “when the probability that the observed components correspond to speech coming from the desired direction (e.g., from the front direction).” Ex. 1006, 10:28–32; *see id.* at 10:40–44. But “if a weak signal or no signal is being received from the front/forward direction, then the value from the adaption control module 502 may be zero.” *Id.* at 10:32–34.

Adaptation processor 504 provides its output (w_0) to equalization module 412. Ex. 1006, 10:53–55, Fig. 5; *see id.* at 10:34–36. Preferably, adaptation processor 504 “adjust[s] the equalization coefficient such that a desired speech signal is cancelled by a backward-facing cardioid pattern.” *Id.* at 10:37–39. Thus, the equalization coefficient “will align a null of a backward-facing cardioid pattern to be directed towards a desired sound source.” *Id.* at 2:23–26; *see id.* at 3:3–5, 5:28–31. Preferably, “the adaptation is performed using a normalized least mean square (NLMS) algorithm having a small step size.” *Id.* at 10:47–49.

2. DIFFERENCES BETWEEN THE CLAIMED INVENTION AND THE PRIOR ART

Claims 32 and 38–40 depend directly or indirectly from claim 29. Ex. 1001, 38:57–61, 39:9–19. As an example, claim 32 depends directly

from claim 29 and requires “applying a delay to the first intermediate signal” where “the delay is proportional to a time difference between arrival of the speech at the second physical microphone and arrival of the speech at the first physical microphone.” *Id.* at 38:57–61. As another example, claim 40 depends indirectly from claim 29 and requires “generating coefficients of the [adaptive] filter during a period when only speech is being received by the first physical microphone and the second physical microphone.” *Id.* at 39:16–19.

For claims 32 and 38–40, Petitioner provides an analysis supported by Dr. Kenny’s testimony about how the combined disclosures in Avendano ’880, Avendano ’252, and Hou teach the claimed subject matter. *See* Pet. 77–91; Ex. 1003 ¶¶ 227–264.

As an example of how Petitioner relies on Avendano ’252, Petitioner asserts that Avendano ’252 teaches claim 32’s requirement for a delay “proportional to a time difference” between speech arrival times at different microphones because Avendano ’252 discloses that:

- (1) “[p]attern generation using differential arrays (DA) requires use of fractional delays whose value may depend on a distance between the microphones”; and
- (2) a delay node or delay line may introduce “a delay equal to an acoustic delay between the primary and secondary microphones 106 and 108.”

Pet. 82–83 (quoting Ex. 1006, 5:36–38, 5:52–54) (citing Ex. 1003 ¶¶ 239–241; Ex. 1006, 5:17–7:64, Figs. 4a–4b, 5).

As another example of how Petitioner relies on Avendano ’252, Petitioner asserts that Avendano ’252 teaches “generating coefficients of the [adaptive] filter during a period when only speech is being received by

the first physical microphone and the second physical microphone”
according to claim 40 because Avendano ’252 discloses that:

- (1) equalization module 412 applies an “adaptive equalization coefficient” as determined by adaptation-control module 502 and adaptation processor 504 “to compensate for microphone mismatch post-manufacturing”;
- (2) adaptation-control module 502 operates as a switch to activate adaptation processor 504 and allow adaptation processor 504 to adjust the “adaptive equalization coefficient”;
- (3) the system may track the “power ratio between the front-facing and back-facing cardioid signals” to “determine if a signal is active in the forward direction, in which case the adaptive equalization coefficient can be updated”;
- (4) adaptation-control module 502 may provide “a value of one” to permit adaptation “when the probability that the observed components correspond to speech coming from the desired direction (e.g., from the front direction)”;
- (5) adaptation-control module 502 may provide a value of zero to prevent adaptation “if a weak signal or no signal is being received from the front/forward direction.”

Pet. 88–90 (emphases omitted) (quoting Ex. 1006, 10:6–11, 10:20–36) (citing Ex. 1003 ¶¶ 260–262; Ex. 1006, 2:17–28, 6:9–38, 10:1–59, 11:56–61, Fig. 5).

Petitioner explains that “when only speech is being received by a first physical microphone and a second physical microphone, the output of a secondary virtual microphone (e.g., a backward-facing cardioid pattern) would be minimized” due to the cancelation of speech from the output.

Pet. 84 (emphasis omitted) (citing Ex. 1003 ¶ 246; Ex. 1006, 10:37–46).

Patent Owner contends that Petitioner has not shown that Avendano '880 and Hou render claim 29 obvious and that Avendano '252 does not remedy this deficiency. Prelim. Resp. 10.

Based on the current record and for the reasons advanced by Petitioner and supported by Dr. Kenny's testimony, Petitioner establishes sufficiently for purposes of institution that the combined disclosures in Avendano '880, Avendano '252, and Hou teach the subject matter in claims 32 and 38–40. *See* Pet. 77–91; Ex. 1003 ¶¶ 227–264.

3. ALLEGED REASONS FOR COMBINING THE TEACHINGS OF THE REFERENCES

Petitioner identifies a reason that would have prompted an ordinarily skilled artisan to combine the teachings of Avendano '880, Avendano '252, and Hou. *See* Pet. 75–76. Additionally, Petitioner explains that an ordinarily skilled artisan “would have had a reasonable expectation of success” in combining the teachings of the references. *Id.* at 76–77. Dr. Kenny's testimony supports Petitioner's positions. *See* Ex. 1003 ¶¶ 220–226.

For instance, Dr. Kenny testifies that an ordinarily skilled artisan “would have recognized” that:

- (1) each of (i) Avendano '880's gain factor “g,” (ii) Hou's filter (e.g., implemented at least in part by Hou's divide unit 512 and/or multiplication unit 514), and (iii) Avendano '252's equalization module 412 serves a “similar purpose, namely compensating for a difference in sensitivity between mismatched microphones to generate directional signals”; and
- (2) Avendano '252's adaption-control module 502 and adaption processor 504 “enhance the signal equalization or compensation process (e.g., by selectively adapting

an ‘equalization coefficient’ of a compensation or equalization filter based on inputs received from the microphones).”

Ex. 1003 ¶¶ 221–222 (emphases omitted) (citing Ex. 1005, 5:36–39, 9:54–59, Figs. 4a–4b; Ex. 1006, 6:30–38, 10:1–59, Figs. 4a–4b, 5; Ex. 1008, 5:26–6:22, 9:28–41, Fig. 5).

Dr. Kenny also testifies that an ordinarily skilled artisan “would have found it obvious to modify” Avendano ’880’s system and/or Hou’s system to include Avendano ’252’s adaption-control module 502 and adaption processor 504 to enhance the performance of Avendano ’880’s gain factor “g” and/or Hou’s filter, e.g., to “enhance the separation of speech and noise in the resulting directional signals.” Ex. 1003 ¶¶ 223–224 (citing Ex. 1006, 10:37–46).

At this stage of the proceeding, Patent Owner makes no arguments against combining the teachings of Avendano ’880, Avendano ’252, and Hou. *See, e.g.*, Prelim. Resp. 10. Nonetheless, the burden remains on Petitioner to demonstrate unpatentability. *See Dynamic Drinkware*, 800 F.3d at 1378.

Based on the current record, Petitioner establishes sufficiently for purposes of institution that an ordinarily skilled artisan would have had a reason, e.g., as articulated by Dr. Kenny, to combine the teachings of Avendano ’880, Avendano ’252, and Hou. *See* Pet. 75–76; Ex. 1003 ¶¶ 220–224. Petitioner also establishes sufficiently for purposes of institution that an ordinarily skilled artisan would have had a reasonable expectation of success in combining the teachings of the references. *See* Pet. 76–77; Ex. 1003 ¶¶ 225–226.

4. PRELIMINARY CONCLUSION ABOUT OBVIOUSNESS/NONOBVIOUSNESS

As discussed above, Petitioner’s analysis addresses how the combined disclosures in Avendano ’880, Avendano ’252, and Hou teach the subject matter in claims 32 and 38–40. *See supra* § IV.E.2. Additionally, Petitioner provides a reason with rational underpinning as to why an ordinarily skilled artisan would have been motivated to combine the teachings of Avendano ’880, Avendano ’252, and Hou and would have had a reasonable expectation of success. *See supra* § IV.E.3. Hence, Petitioner demonstrates a reasonable likelihood of prevailing in proving claims 32 and 38–40 unpatentable under § 103(a) as obvious over Avendano ’880, Avendano ’252, and Hou.

V. CONCLUSION

Based on the arguments and evidence presented by the parties, we determine that there is a reasonable likelihood Petitioner would prevail with respect to at least one claim challenged in the Petition. Hence, we institute an *inter partes* review of all challenged claims on all challenges included in the Petition. *See SAS Inst. Inc. v. Iancu*, 138 S. Ct. 1348, 1355 (2018) (noting that the language of 35 U.S.C. § 314(b) “indicates a binary choice—either institute review or don’t”); *see also PGS Geophysical AS v. Iancu*, 891 F.3d 1354, 1360 (Fed. Cir. 2018) (interpreting the statute as requiring “a simple yes-or-no institution choice respecting a petition, embracing all challenges included in the petition”).

VI. ORDER

Accordingly, it is

ORDERED that, pursuant to 35 U.S.C. § 314(a), an *inter partes* review of claims 29–44 in the ’611 patent is instituted on all challenges included in the Petition;

ORDERED that Petitioner's Motion for Joinder (Paper 3) is denied as moot; and

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

IPR2023-00285
Patent 8,326,611 B2

For PETITIONER:

Colin B. Heideman
Joseph R. Re
Daniel P. Hughes
Nathan D. Reeves
Logan P. Young
KNOBBE, MARTENS, OLSON & BEAR, LLP
2cbh@knobbe.com
2jrr@knobbe.com
2dph@knobbe.com
2ndr@knobbe.com
2lpy@knobbe.com

For PATENT OWNER:

Peter Lambrianakos
Vincent J. Rubino, III
Enrique W. Iturralde
Richard Cowell
FABRICANT LLP
plambrianakos@fabricantllp.com
vrubino@fabricantllp.com
eiturralde@fabricantllp.com
rcowell@fabricantllp.com