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Filed on behalf of Meta Platforms, Inc.

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

META PLATFORMS, INC.,
Petitioner

v.

JAWBONE INNOVATIONS, LLC,
Patent Owner

Case IPR2024-00350
U.S. Patent No. 10,779,080

**Petition for *Inter Partes* Review of
U.S. Patent No. 10,779,080**

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1005	Excerpts of MICROPHONE ARRAYS: SIGNAL PROCESSING TECHNIQUES AND APPLICATIONS (Michael Brandstein & Darren Ward eds., Springer-Verlag 2001) (“Brandstein”)
1006	Lloyd Griffiths & Charles Jim, <i>An Alternative Approach to Linearly Constrained Adaptive Beamforming</i> , vol. AP-30, no. 1 IEEE TRANSACTIONS ON ANTENNAS AND PROPAGATION, 27 (Jan. 1982) (“Griffiths-Jim”)
1007	Japanese Unexamined Patent Application Publication No. H11-18186A to Ikeda et al. (“Ikeda”)
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1009	Iain A. McCowan et al., <i>Near-Field Adaptive Beamformer for Robust Speech Recognition</i> , vol. 12, no. 1 DIGITAL SIGNAL PROCESSING, 87 (Jan. 2002) (“McCowan”)
1010	U.S. Patent Application Publication No. 2002/0193130 to Yang et al. (“Yang”)
1011	U.S. Patent No. 5,651,071 (“Lindemann”)

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1012	U.S. Patent No. 5,627,799 (“Hoshuyama”)
1013	Excerpt of Bernard Widrow & Samuel D. Stearns, <i>Adaptive Signal Processing</i> , Prentice-Hall (1985) (“Widrow”)
1014	Excerpt of Alan V. Oppenheim et al., <i>Discrete-Time Signal Processing</i> , Prentice-Hall, Inc. (2nd ed. 1999) (“Oppenheim”)
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I. INTRODUCTION

Meta Platforms, Inc. (“Petitioner” or “Meta”) requests review under 35 U.S.C. § 311 and cancellation of claims 1-20 of U.S. Patent No. 10,779,080 (EX1001, “the ’080 Patent”) as unpatentable under 35 U.S.C. § 103.

Introduced in the 1980s, the Generalized Sidelobe Canceler (“GSC”) was a foundational noise-reduction technique. Broadly applicable to many signal-processing applications including processing of microphone signals, GSC involves filtering and summing at least two sensor signals in different ways to produce two virtual sensors, one that captures the target plus noise signal and another that captures just noise. Subtracting the noise signal from the target plus noise signal cancels out the noise and yields a noise-reduced output.

In 2001, years before the ’080 Patent’s earliest possible priority date, a widely used reference book, MICROPHONE ARRAYS (EX1005, “Brandstein”), explained it was common to use the GSC with a microphone array to reduce noise in speech signal processing. Meanwhile, other written publications, such as Ikeda, described further noise removal applications using filtering, summing, and linear transfer functions, all of which could be implemented with a GSC.

The challenged claims recite systems that process microphone audio signals to reduce noise. The claimed systems combine two physical microphone signals to form two “virtual” microphones, which must have substantially similar responses to

noise and substantially dissimilar responses to speech. The claimed systems must further include an adaptive noise removal application that performs generic signal-processing operations like filtering and summing the virtual microphone signals, and applying a linear transfer function. These are conventional signal-processing operations and the '080 Patent does not provide any details as to how these generic operations should be applied.

The challenged claims also do not specify how to create the two virtual microphones, other than claim 1's requirement that they are generated by "a processing component." The claims do not limit virtual microphones to those generated using any specific formulas (*compare, e.g.,* EX1001, claim 1 *with id.*, 9:50-63, 11:1). Thus, untethered from any formulas, the claims encompass GSC prior art.

Because they cover techniques published years before the earliest priority date, claims 1-20 are unpatentable. The Board should cancel those claims.

II. BACKGROUND

In 2007, noise reduction techniques had been known for decades, including the prominent GSC, sometimes called Griffiths-Jim beamformer ("GJBF"). (EX1003, ¶¶27-50.)

A. Griffiths-Jim’s Seminal GSC Article

In 1982, Lloyd Griffiths and Charles Jim described the GSC as “a simple time-varying beamformer which can be used to combine the outputs of an array of sensors.” (EX1006, 27, 29.) The beamformer’s purpose was “to minimize the effects of noise and interference at the array output” while capturing the target signal. (*Id.*; EX1003, ¶28)

The GSC (Fig. 4) had two main substructures: the top branch was a “conventional beamformer” designed to capture the target signal plus noise, and the bottom branch was the “sidelobe canceling path” that captured only noise for subtraction or cancellation:

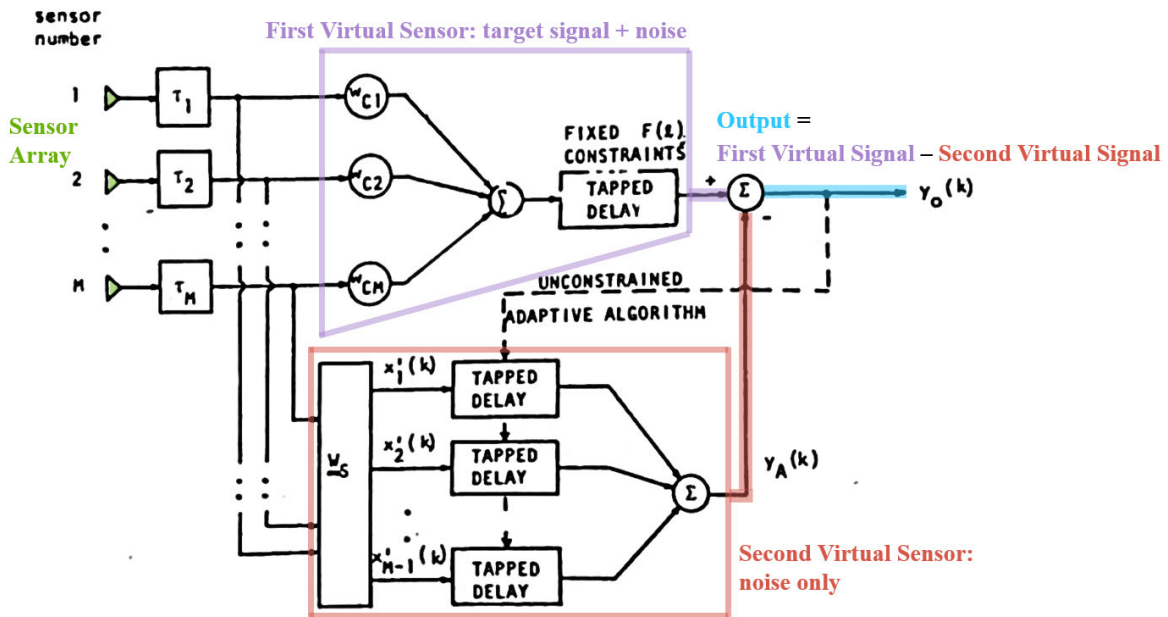


Fig. 4. Generalized sidelobe canceling form of linearly constrained adaptive array processing algorithm.

EX1006, Fig. 4 (annotated)

(EX1006, 29–30¹; EX1003, ¶29.)

The top branch (**first virtual sensor**) combined the **sensor array** outputs, by multiplying them by “weights” ($w_{c1}, \dots, w_{cM}$, or “gains”) and summing the weighted signals, to produce an output $y_c'(k)$ containing the target signal plus noise. (EX1006, 29–30; EX1003, ¶29.)

The bottom branch (**second virtual sensor**) formed a different combination of the **sensor array** outputs by filtering and summing (using a blocking matrix (“BM”) to block the desired signal $s(k)$), to produce an output $y_A(k)$. (EX1006, 30, Fig. 4). The output “ $y_A(k)$ contains no desired signal terms,” “only noise and interference terms.” (*Id.*; EX1003, ¶30.)

The overall **output** $y(k)$ was produced by subtracting the noise only output of the second virtual sensor from the speech and noise output of the first virtual sensor:

$$y(k) = y_c'(k) - y_A(k). \quad (29)$$

(EX1006, 30.) The result was a cleaned-up signal that reduced noise without distorting the desired signal. (EX1003, ¶31; EX1006, 28, 30.)

¹ Figures have been annotated with color throughout.

B. Using GSC to Reduce Noise in Microphone Arrays

In the twenty years following the Griffiths-Jim article, GSC was used in many speech signal processing applications, including with microphone arrays to reduce noise. U.S. Patent No. 5,651,071 (“Lindemann”), filed in 1993, explains that GSC “to improve signal-to-noise ratio for hearing aids” was known. (EX1011, 1:40-46, 12:12–14.²) Griffiths-Jim is also cited in U.S. Patent No. 5,627,799 (“Hoshuyama”), filed in 1995, relating to “interference cancelers, and more particularly to a GSC, or adaptive beamformer for an array of sensors such as microphones[.]” (EX1012, 1:8-11.) Describing the prior art even then, Hoshuyama explains the GJBF was used with microphone arrays. (*Id.*, 1:17-26; EX1003, ¶32.)

C. Brandstein Illustrates Using GSC with Microphone Array to Reduce Noise

Brandstein published in 2001, with its goal to provide “a single complete reference on microphone arrays,” and became a standard reference for audio-signal processing. (EX1005, Preface; EX1003, ¶33.)

Brandstein explains that “[a]pplications of beamforming include microphone arrays for speech enhancement,” which involves “remov[ing] undesirable signals such as noise and reverberation.” (EX1005, 87.) Brandstein further explains that,

² Patent citations are in column:line format.

among various known adaptive beamformers, the GJBF, depicted in Fig. 5.1, “is most widely known.” (*Id.*, 88 (internal citation omitted).)

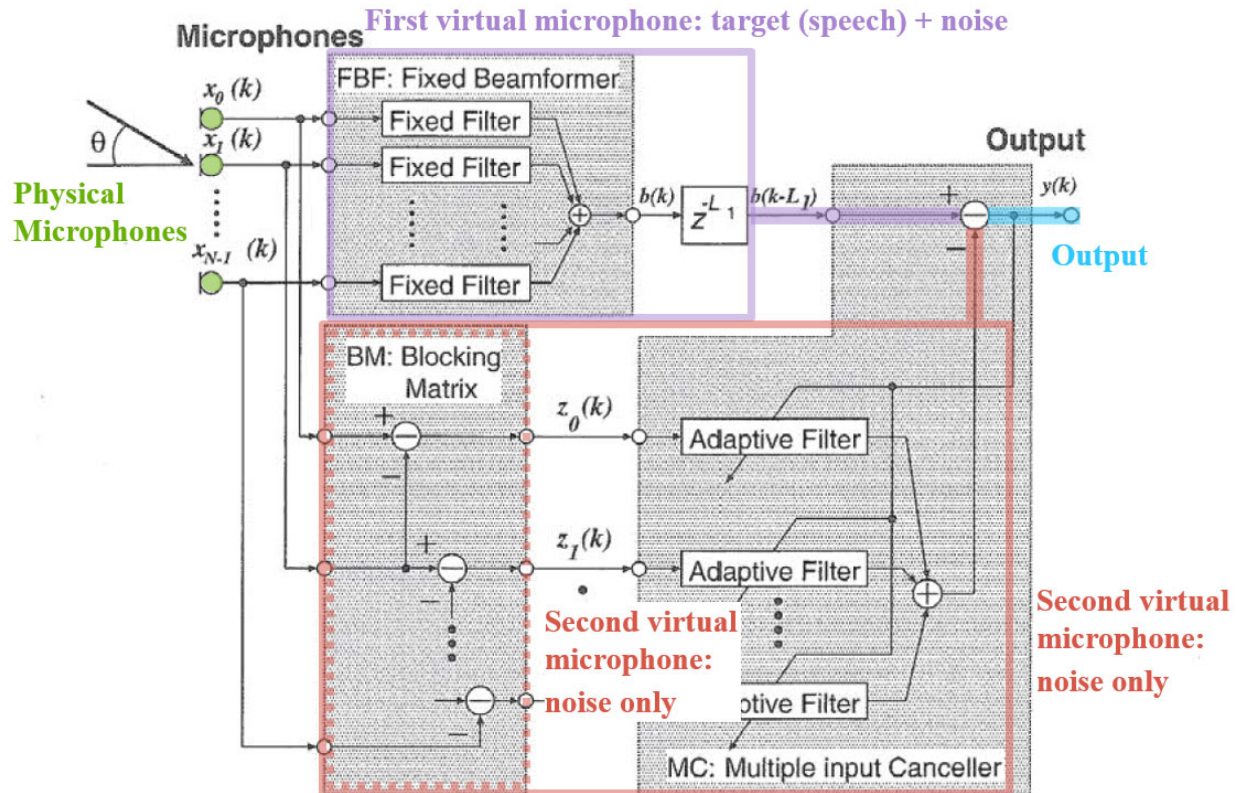


Fig. 5.1. Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceller (MC), and a blocking matrix (BM).

EX1005, 89, Fig. 5.1 (annotated)

A top branch (**first virtual microphone**) combines the signals of **a first and second physical microphone**, $x_0(k)$ and $x_1(k)$, by filtering and summing, forming a fixed beamformer (FBF), capturing target speech plus noise. (EX1003, ¶34.)

The bottom branch (with two options for **second virtual microphone**, as the BM alone or entire bottom branch) combines the two **physical microphone** signals

by summing and filtering in a different way. (*Id.* ¶35.) The **second virtual microphone** captures noise only because it “block[s] the target signal” by “form[ing] a null in the look direction so that the target signal is suppressed and all other signals are passed through.” (EX1005, 88; EX1003, ¶35.)

The overall GJBF **output** is the first virtual microphone’s speech and noise output minus the bottom branch’s noise only output, resulting “in the subtracter output $y(k)$, [for which] the target signal is enhanced and undesirable signals such as ambient noise and interferences are suppressed.” (EX1005, 88-89; EX1003, ¶36.)

The two virtual microphones have very different responses to speech because the first virtual microphone captures the target speech signal, while the second virtual microphone (whether BM only or entire bottom branch) blocks it. (EX1003, ¶37.) But because they have nearly the same responses to noise, the final subtraction removes the noise and enhances the target speech. (*Id.*) This is illustrated in Fig. 5.2, showing the final GJBF output directivity pattern:

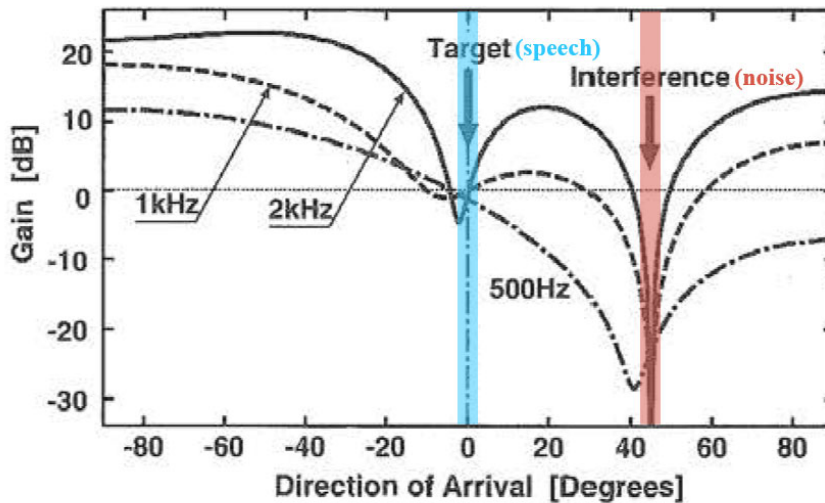


Fig. 5.2. Example directivity pattern of the Griffiths-Jim beamformer.

EX1005, 89, Fig. 5.2 (annotated)

The horizontal axis of the graph shows direction of arrival (“DOA”) in degrees relative to the microphone array: the **target (speech) signal** is at 0°, while the **interference (noise) signal** is shown at approximately 45°. (EX1003, ¶37.) The graph’s vertical axis is gain in decibels: zero gain corresponds to no change in sound pressure or signal power, a positive gain corresponds to an increase in signal power, and a negative gain corresponds to a decrease in signal power. (*Id.*)

As highlighted in **blue**, the target speech signal is reproduced faithfully with essentially zero gain, reflecting that subtracting the bottom branch’s noise only output from the top branch’s speech and noise output produces the speech signal in the speech direction. (*Id.* ¶38.) But, Fig. 5.2 highlights in **red** a highly negative gain

in the interference (*noise*) direction, meaning that subtracting the two branch outputs cancels the noise. (*Id.* ¶¶38-39; *see also* EX1005, 90.)

The basic GJBF (in Fig. 5.1) may have some errors leading to undesirable target “leakage” in the BM and potential cancellation of the target at the output. (EX1005, 90-91, 97.) To reduce this, Brandstein modifies the basic GJBF to track the target’s direction, and adapt the BM coefficients only when the target is absent, to minimize target leakage. *Id.* Brandstein thus incorporates adaptive filters into the GJBF’s bottom branch (containing the *second virtual microphone*), coupled with an adaptation-mode controller (“AMC”) to detect presence or absence of the target signal. *Id.* Brandstein calls this implementation the “GJBF-CNA,” depicted in Fig. 5.8:

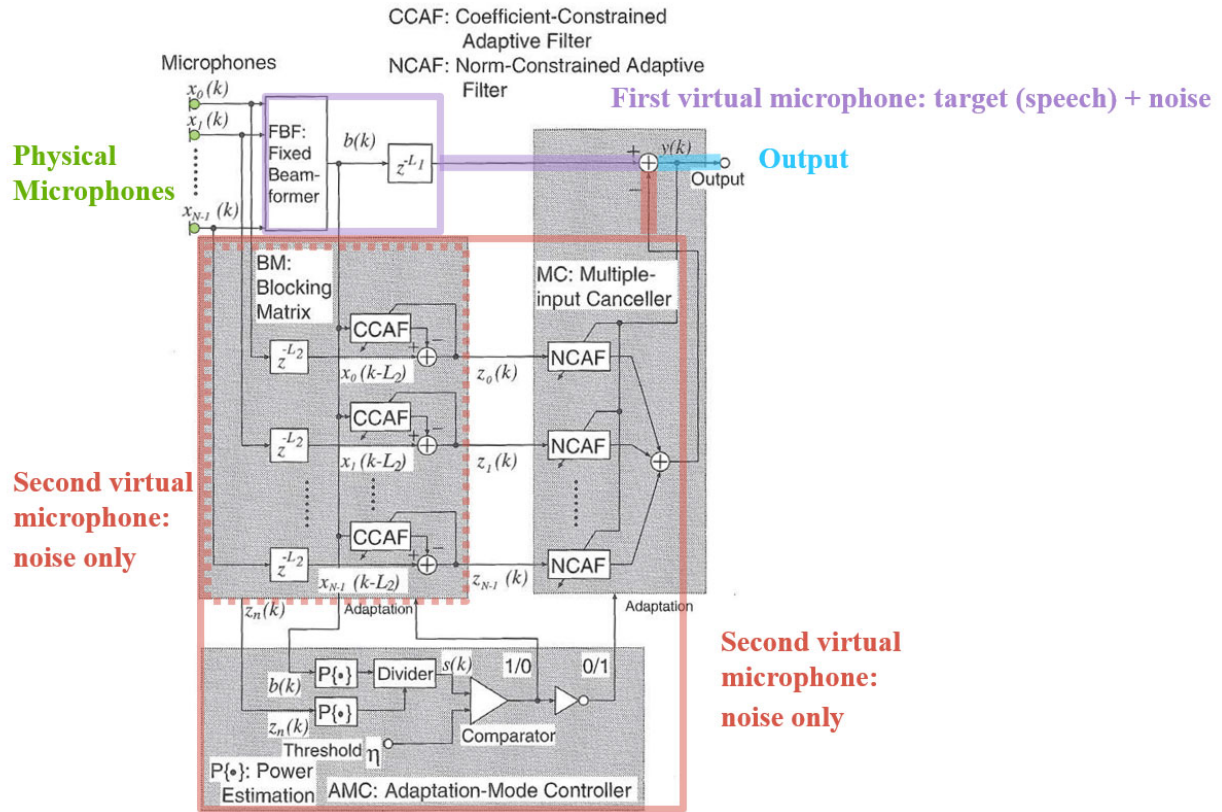


Fig. 5.8. CCAF-NCAF structure with an AMC.

EX1005, 98, Fig. 5.8 (annotated)

(EX1003, ¶¶39-40.)

Brandstein depicts the “sensitivities to the frequency component of the target signal” in Fig. 5.10 (below), for different frequencies. (EX1005, 100.) Because the frequency dependency is small, the GJBF-CNA is suitable for broadband applications such as microphone arrays. (*Id.*; EX1003, ¶41.)

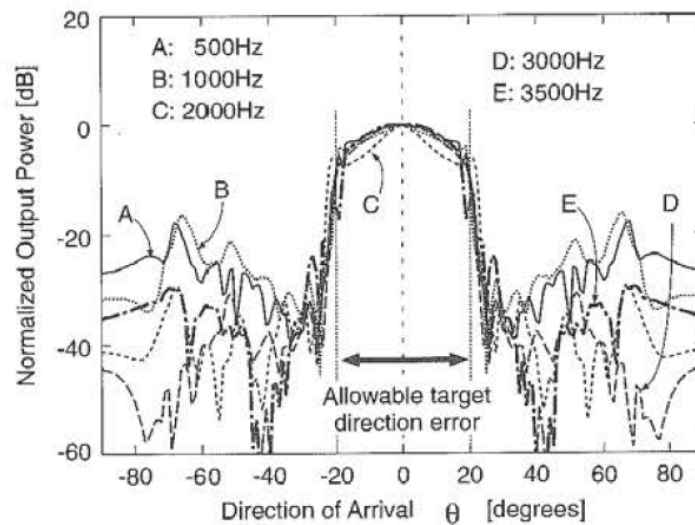
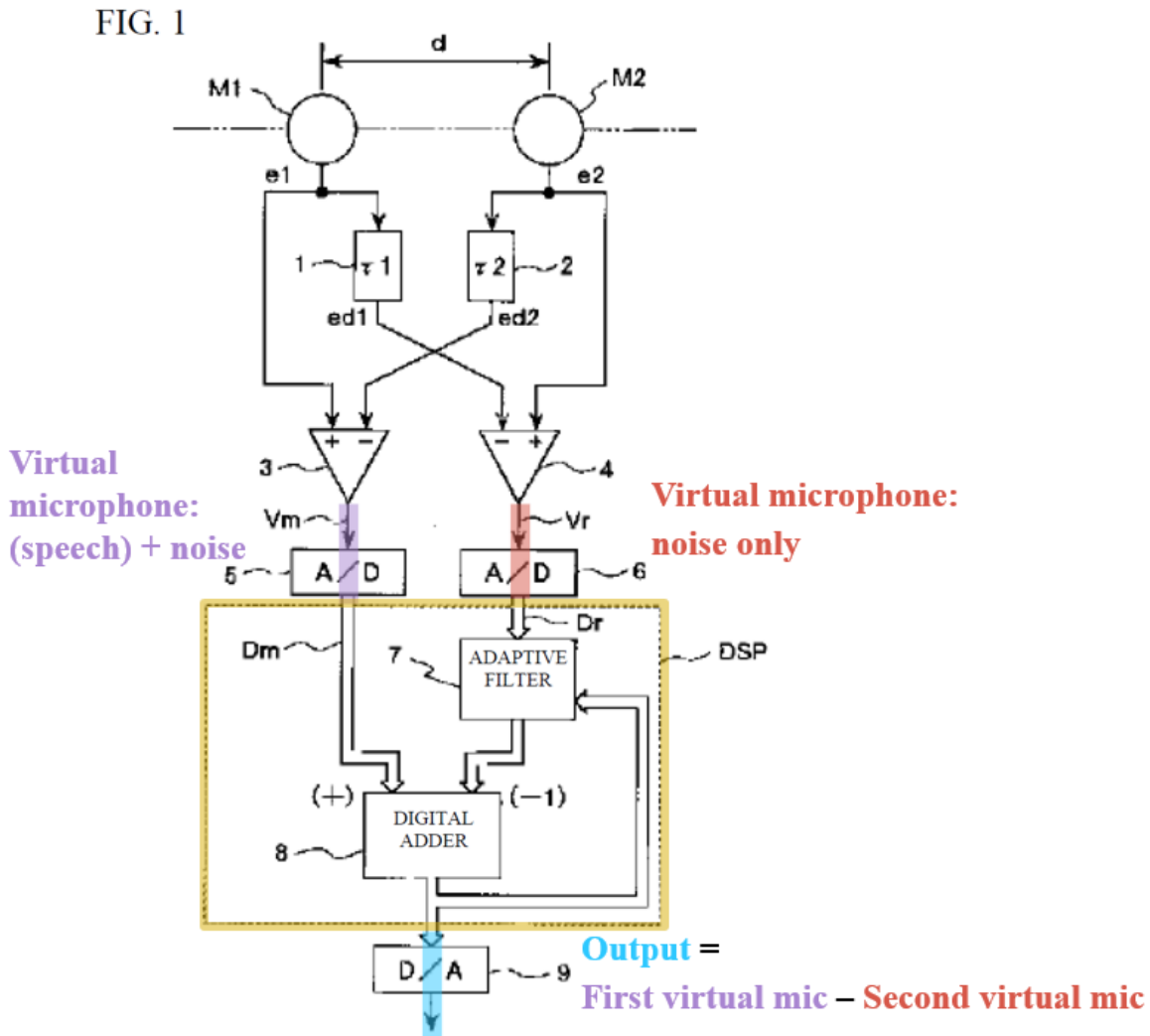


Fig. 5.10. Sensitivities after convergence as a function of DOA at different frequencies.

D. Adaptive Noise Removal Methods Were Well Known

By 2007, it was well-known for adaptive noise removal techniques to include filtering, summing, and transfer functions. For example, Japanese Patent Publication No. H11-18186A (“Ikeda”) discloses a microphone apparatus with two omnidirectional microphones and an adaptive noise cancellation system. (EX1008, ¶¶[0018]-[0024], [0027]-[0030], Abstract, Fig. 1.; EX1003, ¶42.)

Ikeda forms its virtual microphone V_m (main signal) by combining a delayed physical microphone signal M_2 with a physical microphone signal M_1 (EX1008, ¶¶[0002]-[0004], [0012], [0018]-[0022], [0043-44], Figs. 2-3), and another virtual microphone V_r (reference signal) by combining a delayed M_1 with M_2 (EX1008, ¶¶[0012], [0018]-[0020], [0023]-[0024], Figs. 4-5; EX1003, ¶43).



EX1008, Fig. 1 (annotated).

The **main virtual microphone**, like GJBF’s top branch, “has directional characteristics for picking up a target sound and noise,” while the **reference virtual microphone**, like GJBF’s bottom branch, “has directional characteristics for picking up noise.” (EX1008, ¶¶[0021], [0023]; EX1003, ¶44; *see also* EX1008, ¶¶[0012], [0026], Fig. 10, Abstract.)

Ikeda converts V_m and V_r to digital signals D_m and D_r , and inputs them into

a digital signal processor (DSP), which is an adaptive **noise cancellation system** and includes adaptive filter 7 and digital adder 8. (EX1008, ¶¶[0012], [0027]-[0029]; Fig. 1; EX1003, ¶45.) Signal Dr is input to adaptive filter 7, and digital adder 8 subtracts the result from Dm. (EX1008, ¶[0027]; EX1003, ¶45.)

The response characteristics of adaptive filter 7 “are determined so as to cancel the respective noise components contained in” Dm and Dr, resulting in a final **output** “primarily consisting of the target sound component.” (EX1008, ¶¶[0028]-[0029], [0012], Abstract; EX1003, ¶46.)

E. Delaying Signals Based on Geometry to Adjust for Differences in Arrival Times Was Well-Known

Adaptive beamforming relies on the array’s spatial geometry and the signal’s DOA. (EX1003, ¶47; EX1005, 87.) When the target source is close enough to the array, instead of treating the signal’s wavefront as a plane, it is useful to use a *near-field* design accounting for the wavefront’s spherical geometry. (EX1003, ¶¶47.) Near-field designs use basic calculations to determine the delays between signal arrival at various microphones. (*Id.*) Brandstein discloses:

- Nearfield design, including attenuation of the desired signal [14], [22]

$$\mathbf{d}^T = [a_0 \exp(-j\omega\tau_0), a_1 \exp(-j\omega\tau_1), \dots, a_{N-1} \exp(-j\omega\tau_{N-1})], \quad (2.28)$$

$$a_i = \frac{\|\mathbf{q} - \mathbf{p}_{ref}\|}{\|\mathbf{q} - \mathbf{p}_i\|}, \quad (2.29)$$

$$\tau_i = \frac{\|\mathbf{q} - \mathbf{p}_{ref}\| - \|\mathbf{q} - \mathbf{p}_i\|}{c}, \quad (2.30)$$

where $\|\mathbf{q} - \mathbf{p}_{ref}\|$ and $\|\mathbf{q} - \mathbf{p}_i\|$ denote the distance between the vector location of the source $\mathbf{q}$ and a reference sensor $\mathbf{p}_{ref}$, or the sensor $\mathbf{p}_i$, respectively.

(EX1005, 25.) A signal's travel time from source to microphone is the distance traveled divided by c , the speed of sound. (EX1003, ¶47.) Brandstein's τ_i thus represents the difference in time for the signal to propagate from source to microphone i , relative to that for a reference microphone. (*Id.*) Including the τ_i term in the complex exponential shown above and applying that exponential to the reference microphone signal represents delaying the received signal to reproduce (in terms of phase) the signal as received at microphone i . (*Id.*) Applying the a_i term to the reference signal adjusts the amplitude of the signal to match the amplitude of the signal as it is received at microphone i . (*Id.*) In other words, the signal x_i on microphone i can be expressed in terms of the signal x_{ref} on the reference microphone as follows (where $j = \sqrt{-1}$):

$$x_i(\omega) = a_i(e^{j\omega\tau_i})x_{ref}(\omega).$$

(*Id.*)

The reasoning behind these equations is further explained in a 2002 article by McCowan, which expressly motivates and teaches incorporating near-field design into GSC. (EX1009, 89-96; EX1003, ¶48.)

F. GSCs Incorporated Voice Activity Detection

GJBFs incorporated voice activity detection, which was known by 2007. Yang teaches a noise suppression system that comprises a GJBF and an adaptive noise suppression unit. (EX1010, ¶¶[0026], [0038], [0058]-[0062], Fig. 4; EX1003, ¶49) Within the adaptive noise suppression unit, there is a **voice activity detector** (“VAD”) 440 and an adaptive filter 450 that adapts filter coefficients when there is no speech activity. (EX1010, ¶¶ [0060]-[0061]; EX1003, ¶50.)

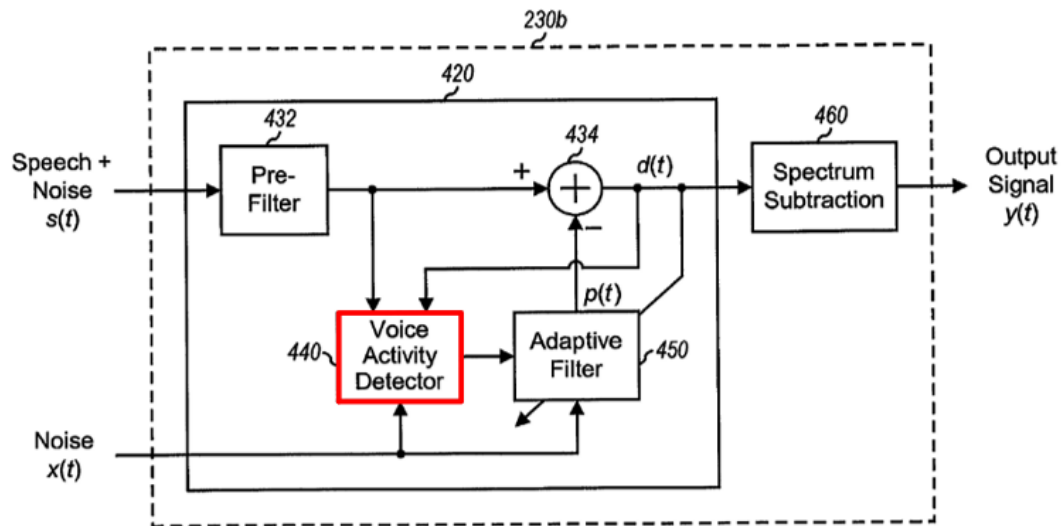


FIG. 4

EX1009, Fig. 4 (annotated).

III. THE '080 PATENT

A. Specification

The '080 Patent purports to distinguish from “conventional arrays and algorithms” by using a virtual microphone with a null in the speech direction “to remove the speech of the user.” (EX1001, 3:49-57.) But this is exactly what the GSC does. (EX1003, ¶¶51-56.)

Specifically, the '080 Patent describes adaptive noise suppression. (EX1001, 3:49-50, 4:52-53, Abstract.) “[T]wo distinct virtual directional microphones... have very similar noise responses and very dissimilar speech responses” (*id.*, 3:49-56, 4:48-51, 7:39-51), and “pair[] with an adaptive filter algorithm” to reduce noise. (*Id.*, 3:57-62; EX1003, ¶52.)

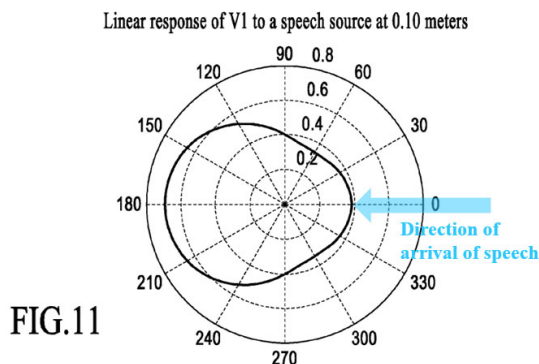
In one embodiment, two virtual microphones V1 and V2 are formed based on two physical microphone outputs O1 and O2, based on certain parameters:

$$\begin{aligned}\beta &= \frac{d_1}{d_2} \\ \gamma_- &= \frac{d_2 - d_1}{c} \cdot f_x \text{ (samples)} \\ d_1 &= \sqrt{d_S^2 - 2d_{S1}d_0\cos(\theta) + d_0^2} \\ d_2 &= \sqrt{d_S^2 + 2d_Sd_0\cos(\theta) + d_0^2}\end{aligned}$$

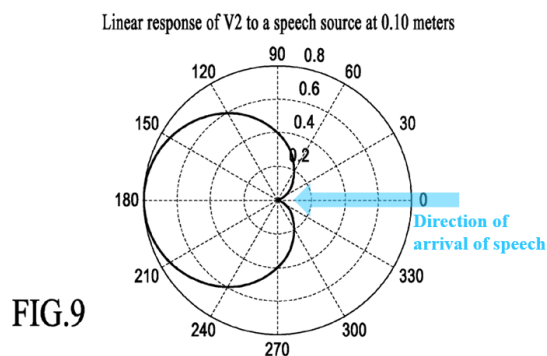
(*Id.*, 9:55-62.) “The distances d_1 and d_2 are the distance from O_1 and O_2 to the speech source (see FIG. 2), respectively, and γ is their difference divided by c , the speed of

sound, and multiplied by the sampling frequency f_s .” (*Id.*, 9:64-67.) Thus, γ is the difference in propagation time to the microphones from the standard near-field design equations (denoted τ_i in §II.E, *supra*), expressed in samples. Similarly, β is the amplitude-adjustment factor from the standard near-field design (denoted a_i in §II.E, *supra*). (EX1003, ¶53.)

According to the ’080 Patent, V_1 and V_2 have desirable directivity patterns with respect to a speech signal expected to arrive at an angle of 0° :



EX1001, Fig. 11 (annotated)



EX1001, Fig. 9 (annotated)

As shown, V_1 (Fig. 11) exhibits a non-zero speech response, while V_2 (Fig. 9) has a null in the speech direction. (EX1001, 10:18-30, 11:20-24.) In contrast, V_1 and V_2 have “very similar” responses to noise. (*Id.*, 11:25-29; EX1003, ¶54.) For different frequencies within -30 to 30° , the V_1 response is flat, as shown in Fig. 13 (reproduced below). (EX1001, 11:30-35, 11:43-61; EX1003, ¶55.)

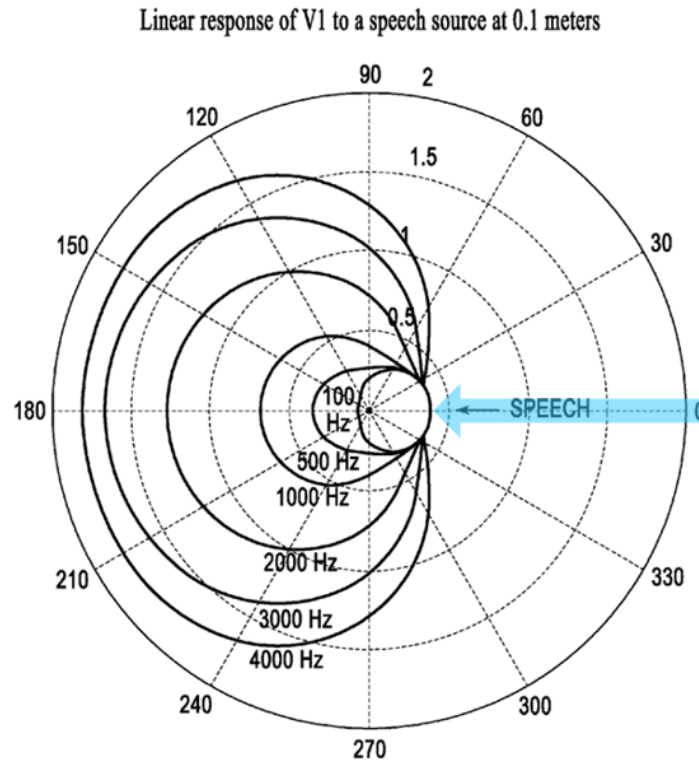


FIG.13

EX1001, Fig. 13 (annotated).

The virtual microphone signals are input to an adaptive noise removal system, which can use “any adaptive filter or noise removal algorithm.” (EX1001, 3:57-62, 4:52-7:16, 8:50-9:3, 9:22-38; EX1003, ¶56.) The adaptive noise removal system combines V_1 and V_2 signals “by filtering and summing in the time domain” and using a first microphone signal “to remove noise from the speech received from” another microphone signal, which “relies on a slowly varying linear transfer function between the two microphones for sources of noise.” (EX1001, 7:4-16, 8:50-9:3, 9:22-38.) The system output’s “noise content is attenuated with respect to the speech

content.” (EX1001, 7:12-16.)

B. Prosecution History

The '080 Patent was filed on July 22, 2013, as U.S. Patent Application No. 13/948,160 (“the '160 application”), claiming priority to various applications, the earliest being filed on June 13, 2007. (EX1003, ¶57.)

During prosecution, the Examiner rejected the claim as anticipated by U.S. Patent No. 8,068,619 and double patenting. (EX1002, 57, 98-106.) In response, the applicant canceled the claim and added twenty new claims (*id.*, 113-123) and filed a terminal disclaimer (*id.*, 142-153, 160-168, 222-234, 248). In allowing the claims, the Examiner stated “a dual virtual microphone array as claimed including having a linear response to both speech and noise could not be anticipated nor, in combination, be rendered obvious over the prior art of record.” (*Id.*, 249-256; EX1003, ¶58)

The Examiner did not have the benefit of any of the prior art asserted in the grounds of this Petition, which discloses the claimed features that Applicant argued distinguished the prior art during prosecution and that the Examiner referenced in allowing the claims.

IV. STATEMENT OF RELIEF REQUESTED

A. Grounds

The Board should cancel the claims as follows:

Ground	Claims Challenged	References
1	1-3, 5-9, 11-14, 16-20	Brandstein, Ikeda
2	13, 19, 20	Brandstein, Ikeda, McCowan
3	4, 10, 15	Brandstein, Ikeda, Yang

B. Priority Date

The earliest application to which the '080 Patent claims priority is U.S. provisional application 60/934,551, filed June 13, 2007, which is therefore the earliest possible priority date for all challenged claims. Petitioner does not concede that the claims are entitled to that priority date. (EX1003, ¶59.)

C. The References Are Prior Art

All references relied on in the Grounds are prior art under pre-AIA §102(b) because they published more than one year before June 13, 2007:

- (1) Brandstein published no later than 2001 (EX1005; EX1022; EX1003, ¶60);
- (2) Ikeda published no later than 1999 (EX1007; EX1008; EX1003, ¶60);
- (3) McCowan published no later than 2002 (EX1009; EX1022; EX1003, ¶60);
- (4) Yang published no later than 2002 (EX1010; EX1003, ¶60).

D. The Asserted References Are Analogous Art

Prior art is analogous if it is from the same field of endeavor, regardless of the problem addressed, or it is reasonably pertinent to the particular problem with which the inventor is involved. *Unwired Planet, LLC v. Google Inc.*, 841 F.3d 995, 1000 (Fed. Cir. 2016). Petitioner’s references are analogous because they are from the same field as the ’080 Patent, *e.g.*, microphone array signal processing and noise reduction. (EX1003, ¶¶61; EX1001, 1:21-24, 3:49-50.) They are also pertinent to a problem the inventor was focused on, *e.g.*, noise suppression. (EX1003, ¶¶61; EX1001, 1:21-24, 3:49-50.)

V. LEVEL OF ORDINARY SKILL

Based on the relevant factors, *In re GPAC Inc.*, 57 F.3d 1573, 1579 (Fed. Cir. 1995), a person of ordinary skill in the art (“POSITA”) would have had a minimum of a bachelor’s degree in computer engineering, computer science, electrical engineering, mechanical engineering, or a similar field, and approximately three years of industry or academic experience in a field related to acoustics, speech recognition, speech detection, or signal processing. (EX1003, ¶¶21-26.) Work experience could substitute for formal education, and additional formal education could substitute for work experience. (*Id.*)

VI. CLAIM CONSTRUCTION

On February 28, 2023, Patent Owner (“PO”) sued Petitioner for infringement of the ’080 Patent in *Jawbone Innovations, LLC v. Meta Platforms, Inc.*, No. 6:23-cv-00158 (W.D. Tex. 2023). Although the parties may propose constructions for certain terms in the Litigation for purposes of determining infringement, no claim terms require express construction to resolve the obviousness challenges here.³ *Nidec Motor Corp. v. Zhongshan Broad Ocean Motor Co. Ltd.*, 868 F.3d 1013, 1017 (Fed. Cir. 2017); *Vivid Techs., Inc. v. Am. Sci. & Eng’g, Inc.*, 200 F.3d 795, 803 (Fed. Cir. 1999).

³ In district court, the parties may dispute whether some claims are invalid under 35 U.S.C. § 112. During claim construction in *Jawbone Innovations, LLC v. Google LLC*, C.A. No. 21-cv-00985 (W.D. Tex. Sept. 23, 2021), the Court determined that the terms “substantially similar” and “substantially dissimilar” are indefinite. (EX1021.) For this proceeding only, Petitioner assumes the claims are not invalid under § 112 and interprets “substantially similar” responses to include nearly identical responses, and “substantially dissimilar” responses to include very different responses.

VII. GROUNDS OF UNPATENTABILITY

A. Ground 1: Brandstein in View of Ikeda Render Obvious Claims 1-3, 5-9, 11-14, and 16-20

Petitioner relies on two embodiments in Brandstein for a modified GJBF structure, and Ikeda's teaching of adaptive noise cancellation. Brandstein's basic GJBF (Figs. 5.1-5.3) meets nearly all limitations of the independent claims, except for the limitations related to (1) the *first linear response to speech being substantially similar across a plurality of frequencies* and (2) the *adaptive noise removal application*. (EX1003, ¶¶62-77.)

For (1), Petitioner relies on a modification to the basic GJBF, which Brandstein calls a GJBF-CNA (Figs. 5.8-5.10), which Brandstein teaches has a *response to speech* that is constant *across a plurality of frequencies*. When modifying the basic GJBF, the POSITA would have understood that the overall characteristics and shape of the response patterns would remain unchanged, and therefore would have also applied to the GJBF-CNA. (EX1003, ¶¶64.)

For (2), Petitioner relies on an additional stage of *adaptive noise removal* (taught in Ikeda) following the modified GJBF, to remove residual single-channel noise remaining after the beamformer. (EX1003, ¶¶64.)

1. Motivation to Combine

a. Brandstein’s Different Embodiments

POSITA would have been motivated to modify Brandstein’s basic GJBF with Brandstein’s GJBF-CNA structure, so that the *first linear response to speech is substantially similar across a plurality of frequencies*. (EX1005, 99-100 (describing the response as “constant” for different frequencies), Fig. 5.10; EX1003, ¶64.)

Brandstein teaches that target signal cancellation due to steering-vector errors may occur in the basic GJBF, which leads to undesirable target “leakage” in the BM and target signal cancellation at the output. (EX1005, 90-91, 97.) To alleviate this, Brandstein’s GJBF-CNA incorporates adaptive filters into the bottom branch, coupled with an AMC to detect presence or absence of the target signal. (*Id.*, 87-90, Fig. 5.8.) With these modifications, the modified GJBF tracks the direction of the target signal, and adapts the BM coefficients only when the target signal is absent, to minimize target leakage and therefore target cancellation. (*Id.*; EX1003, ¶¶65-66.) Moreover, because both Brandstein embodiments share the same goals and have the same overall beamforming structure (*e.g.*, subtracting a noise only signal from a target plus noise signal using spatial channels), the POSITA would have understood that the responses’ overall shapes and characteristics for the basic GJBF’s top and bottom branches (including the FBF and BM) remain unchanged for the modified GJBF. (EX1003, ¶67.) Thus, a POSITA would have been motivated

to apply Brandstein’s teachings Brandstein’s GJBF-CNA to the basic GJBF. (*Id.*, ¶¶68-70.)

b. Brandstein and Ikeda

While Brandstein generally discloses noise removal (EX1005, 88-90), it does not provide explicit details regarding their techniques. Specifically, Brandstein’s modified GJBF reduces spatial noise by beamforming, but does not explicitly reduce the residual single-channel noise. Ikeda provides this detail, and the POSITA would have been motivated to implement Ikeda’s removing of residual single-channel noise leftover after Brandstein’s beamforming. (EX1003, ¶71.)

Ikeda performs adaptive noise removal on signals similar to Brandstein’s top and bottom branch signals of the modified GJBF (*first and second virtual microphone signals*, respectively). It would have been obvious to implement Ikeda’s DSP after Brandstein’s modified GJBF because Brandstein’s top branch output is similar to Ikeda’s main signal *V_m*, as both contain speech and noise. (*Id.*)

For the *second virtual microphone*, Petitioner presents two alternative mappings for the combination of Brandstein with Ikeda and explains the motivations for each mapping separately below. (*Id.*, ¶72.)

In a first mapping, a POSITA would have been motivated to apply Ikeda’s DSP to the output of Brandstein’s MC’s summer (together with the BM and the MC’s adaptive filters, the *second virtual microphone*). (*Id.*, ¶73.) The result is that

Brandstein's MC summer output is the reference signal V_r for feeding into Ikeda's DSP.

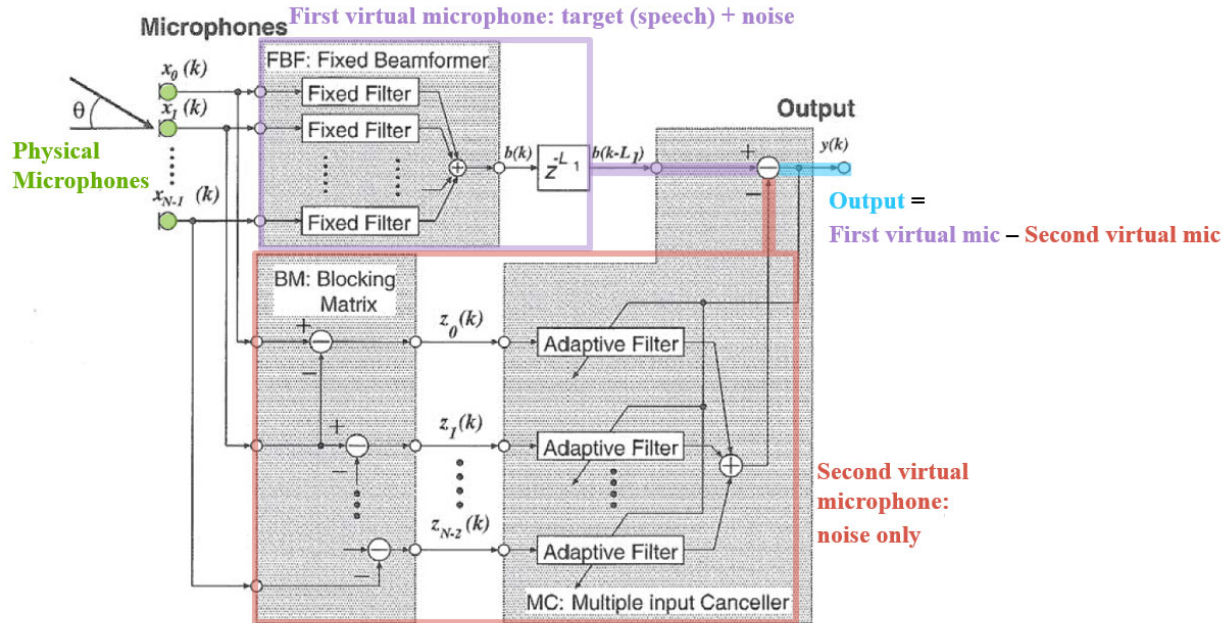


Fig. 5.1. Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceller (MC), and a blocking matrix (BM).

EX1005, 89, Fig. 5.1 (annotated)

A POSITA would have recognized that adding Ikeda's DSP to Brandstein's modified GJBF in this manner would have improved the noise-cancellation performance of the modified GJBF. (*Id.*, ¶74.) Specifically, the POSITA would have known that including additional noise-cancellation stages would have been beneficial as it would have improved system robustness and stability, while also reducing residual single-channel noise and distortion remaining from the beamformer, which mainly reduces spatial noise across multiple channels. (*Id.*)

In a second mapping, a POSITA would have been motivated to apply Ikeda’s DSP to the output of Brandstein’s BM (the *second virtual microphone*). (*Id.*, ¶75.) The result is that Brandstein’s BM output is the reference signal V_r for feeding into Ikeda’s DSP. Specifically, Brandstein’s MC has a similar structure as Ikeda’s DSP, as both include adaptive filters that receive feedback from a noise-reduced output signal, and include a difference module that subtracts the filtered signal (a noise signal) from a speech and noise signal. (*Id.*) As shown below, the BM’s output passes through the MC’s adaptive filter, much like how Ikeda’s reference signal V_r passes through the DSP’s adaptive filter 7 (*compare* Brandstein’s MC in Fig. 5.1 *with* Ikeda’s DSP in Fig. 1):

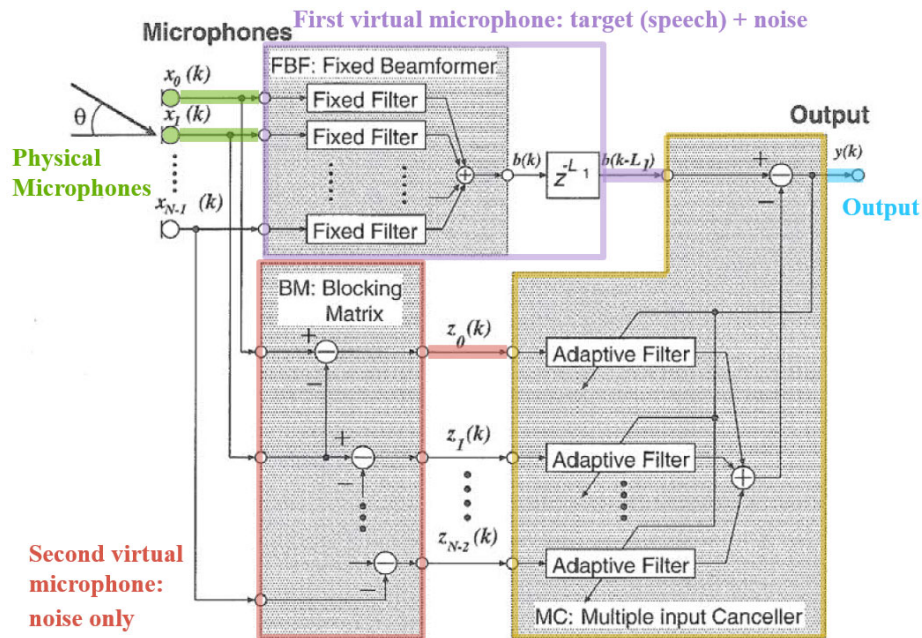
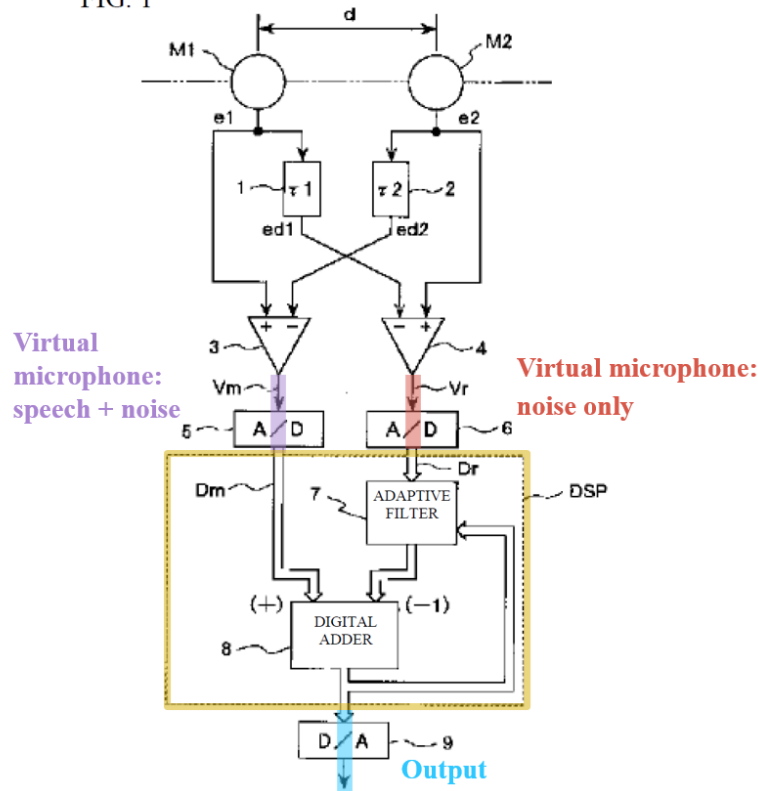


Fig. 5.1. Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceler (MC), and a blocking matrix (BM).

EX1005, 89, Fig. 5.1 (annotated)

FIG. 1



EX1008, Fig. 1 (annotated).

In Brandstein, the output of the MC's summer is subtracted from the top branch output to produce overall output $y(k)$, which is provided as feedback to the MC's adaptive filter. Likewise, Ikeda's DSP subtracts the output of the adaptive filter 7 from the main signal V_m (which, like the FBF signal of the GSC, contains speech and noise), to produce an overall output that is fed back into the adaptive filter 7. Given these structural similarities, a POSITA would have been motivated to apply the specific teachings of Ikeda's DSP to Brandstein's MC. (*Id.*, ¶76.)

For these reasons, a POSITA would have known that implementing Brandstein’s GSC to include Ikeda’s DSP would have further advanced Brandstein’s goal of removing undesirable noise and reverberation signals. (*Id.*, ¶77.)

2. Reasonable Expectation of Success

a. Brandstein’s Different Embodiments

A POSITA would have had a reasonable expectation of success to modify the basic GJBF, while maintaining the general characteristics of the GJBF, FBF, BM, and top and bottom branch directivity patterns for the modified GJBF. (*Id.*, ¶¶78-83.)

As a POSITA would have recognized, the GJBF-CNA operates on the same principles as the GSC (subtracting the noise only output of the *second virtual microphone* from the speech and noise output of the *first virtual microphone*). A POSITA would have further understood that the directivity pattern of the final output, FBF, and BM would have the same features as that depicted in the Figs. 5.2 and 5.3 directivity plots. (*Id.*, ¶¶78-79.) As shown in Fig. 5.8, the FBF is unchanged in the GJBF-CNA. (EX1005, 98; EX1003, ¶79.) Therefore, the response of the FBF in the GJBF-CNA would be identical to the FBF in the basic GSC, and the FBF output would still contain speech and noise. (EX1003, ¶¶79-80.) Brandstein also teaches that the BM in the GJBF-CNA “provides a wider null for the target with sharper edges.” (EX1005, 97.) Other than widening the null, the POSITA would

have understood that the GJBF-CNA bottom branch does not otherwise modify the overall shape and characteristic of the response pattern from the BM response shown in Fig. 5.3. (EX1003, ¶81.) This is because the added adaptive filters in the GJBF-CNA bottom branch provide improved tracking to movements of the speech source, but do not change the overall goal of the GJBF to cancel the noise. (*Id.*) Therefore, a POSITA would understand that the BM's response would maintain relevant characteristics, such as having a single null in the speech response toward the direction of speech. (*Id.*, ¶¶82-83.)

b. Brandstein and Ikeda

A POSITA would have expected to be able to successfully implement Ikeda's DSP in or after Brandstein's modified GJBF, as the two structures are compatible. (*Id.*, ¶¶84-85.) Specifically, Ikeda's main virtual microphone V_m (like the GJBF top branch) picks up target and noise, while its reference virtual microphone V_r (like the GJBF bottom branch) picks up just noise. (EX1008, ¶¶[0021], [0023]; EX1003, ¶84; *see also* EX1008, ¶¶[0012], [0026], Fig. 10, Abstract.) Therefore, the virtual microphone signals of Brandstein are akin to the virtual microphone signals received by the DSP of Ikeda.

As discussed above in Section VII.A.1.b, both virtual microphones are input to Ikeda's DSP. (EX1008, ¶¶[0021], [0023]; EX1003, ¶85.) As the POSITA would have recognized, processing of noise only or speech and noise signals for further

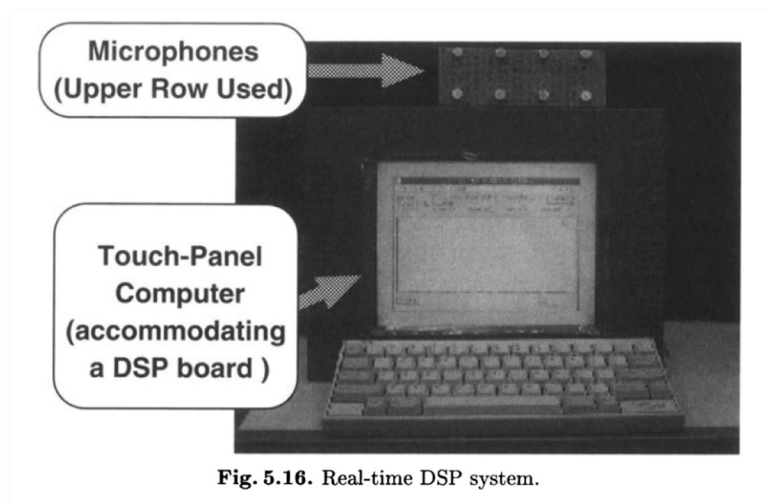
noise reduction or noise cancellation from Ikeda would have been readily applied to Brandstein, and the POSITA would have had a reasonable expectation of success in applying Ikeda’s DSP to Brandstein’s GJBF. (EX1003, ¶¶85.)

3. Independent Claim 7

Petitioner starts its analysis with independent claim 7, then addresses independent claims 14 and 1. (EX1003, ¶¶86-144.)

a. [7pre] “A system, comprising:”

To the extent the preamble is limiting, Brandstein discloses this limitation. (EX1003, ¶¶87-88.) Brandstein discloses a *system* with a microphone array and signal processing that implements the GJBF (EX1005, 89, Figs. 5.1, 5.8, 5.16.), and reports GJBF experimental results on a “Real-time DSP *system*,” illustrated in Fig. 5.16:



EX1005, 105, Fig. 5.16 (annotations in original)

Specifically, the *system* has a microphone array and a computer. (*Id.*, 104.)

Moreover, a POSITA would have understood that Brandstein’s GJBF would have been implemented in a *system* for beamforming, such as in telephones or hearing aids. (*Id.*, 87; EX1003, ¶¶88.)

- b. [7a] “a first virtual microphone formed from a first combination of a first microphone signal and a second microphone signal, wherein the first microphone signal is generated by a first physical microphone and the second microphone signal is generated by a second physical microphone;”

Brandstein discloses this limitation. (EX1003, ¶¶89-90.)

Fig. 5.1 shows the output $b(k-L_1)$ of the basic GJBF’s top branch (including the FBF and a delay module, together corresponding to the *first virtual microphone*⁴), which is the delayed summation of filtered *first and second microphone signals* (e.g., x_0 and x_1) output (generated) by *first and second physical microphones*, respectively. Because the filtered signals are summed, the *first virtual microphone* is *formed* by combining the microphone signals (*a first combination*):

⁴ Petitioner maps the *first virtual microphone* to the GJBF’s top branch, which includes an FBF and a delay module. The POSITA would have understood that the response pattern for the FBF is the same as that for the entire top branch, as it is unaffected by the delay module. (EX1003 ¶¶67, 70.)

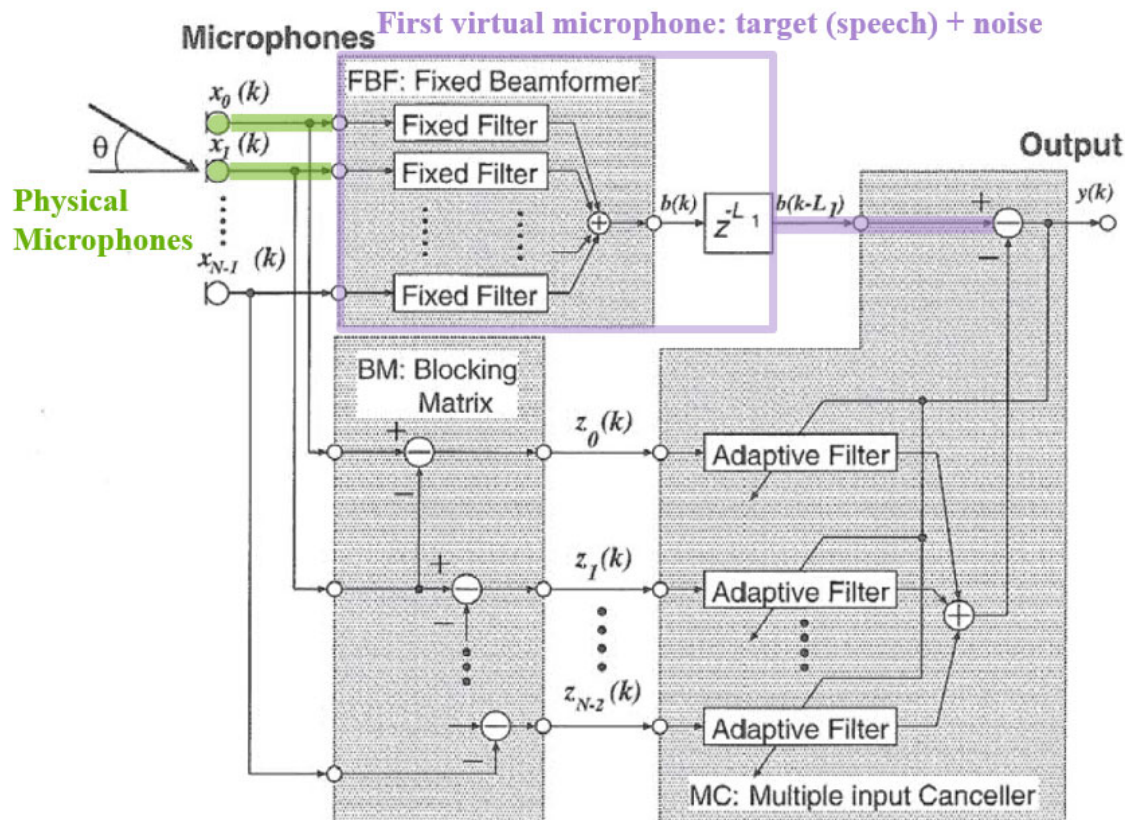


Fig. 5.1. Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceller (MC), and a blocking matrix (BM).

EX1005, 89, Fig. 5.1 (annotated)

(*Id.*, ¶90.) Although not explicitly required in [7a], Brandstein also discloses the physical microphones are omnidirectional. (EX1005, 101; EX1003, ¶90.)

- c. [7b] “a second virtual microphone formed from a second combination of the first microphone signal and the second microphone signal, wherein the second combination is different from the first combination,”

Brandstein discloses this limitation. (EX1003, ¶¶91-94.)

In the first mapping, *the second virtual microphone*⁵ signal is the bottom branch's output. (*Id.*, ¶¶91-93.) As shown in Fig. 5.1, a *second virtual microphone* along the GJBF's bottom branch is *formed from* a *second combination* of the *first and second microphone signals* (e.g., x_0 and x_1) that *is different from the first combination* because the bottom branch uses subtractors and adaptive filters instead of fixed filters (*Id.*, ¶93):

⁵ In this mapping, Petitioner maps the *second virtual microphone* to the GJBF's bottom branch, which includes a BM and adaptive filters and a summer module in the MC. The POSITA would have understood that the response pattern for the BM is the same as that for the entire bottom branch, as its overall shape and characteristics are not significantly affected by the MC components. (EX1003 ¶93.)

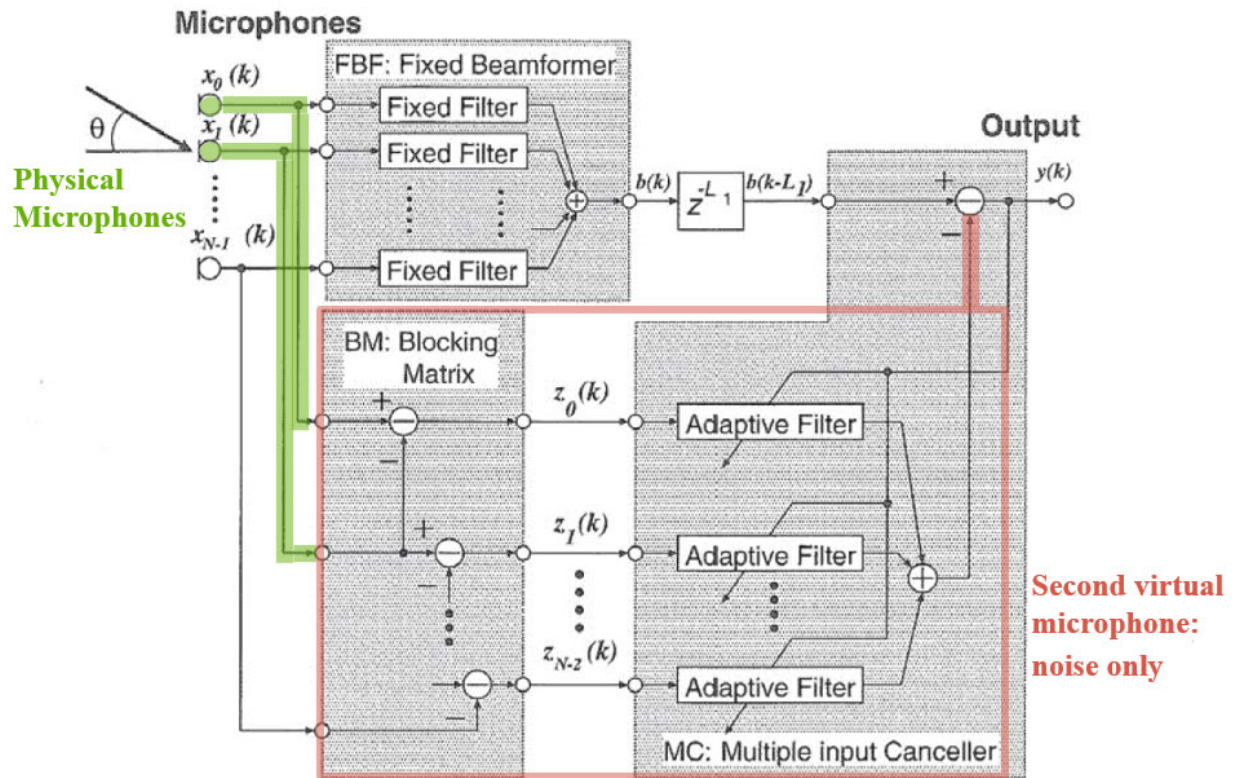


Fig. 5.1. Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceler (MC), and a blocking matrix (BM).

First Mapping, EX1005, 89, Fig. 5.1 (annotated)

In the second mapping, the *second virtual microphone* signal is the BM's output. Here, the *second virtual microphone* is the BM, which is formed from a *second combination* of the *first and second microphone signals* (x_0 and x_1) that is *different from the first combination* because the BM uses subtractors instead of fixed filters (*Id.*, ¶94):

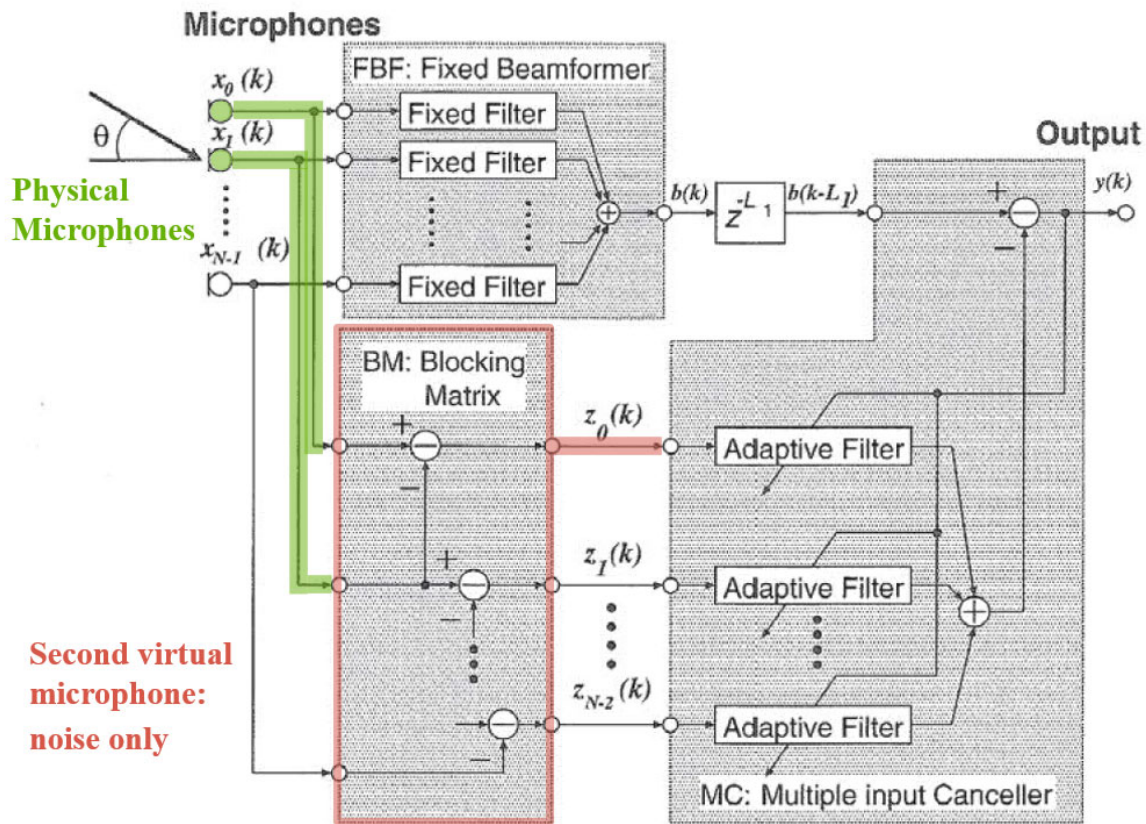


Fig. 5.1. Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceller (MC), and a blocking matrix (BM).

Second Mapping, EX1005, 89, Fig. 5.1 (annotated)

- d. [7c] “wherein the first virtual microphone has a first linear response to speech and first linear response to noise,”

Brandstein discloses this limitation. (*Id.*, ¶¶95-100.)

Brandstein’s *first and second virtual microphones* (top and bottom GJBF branches, respectively) output separate signals. As discussed in Section VII.A.3.b ([7a]), the GJBF top branch includes an FBF and corresponds to the *first virtual microphone*. (*Id.*, ¶96.) The target signal is *speech*, positioned at 0° (EX1005, 101 (“The target source was an English male speech signal.”); *also id.*, 103, Figs. 5.13,

5.14), while the interference signal is *noise*,⁶ positioned at approximately 45°. (EX1003, ¶¶96-97).

The top plot of Fig. 5.3 shows the FBF's directivity pattern (*Id.*, ¶98):

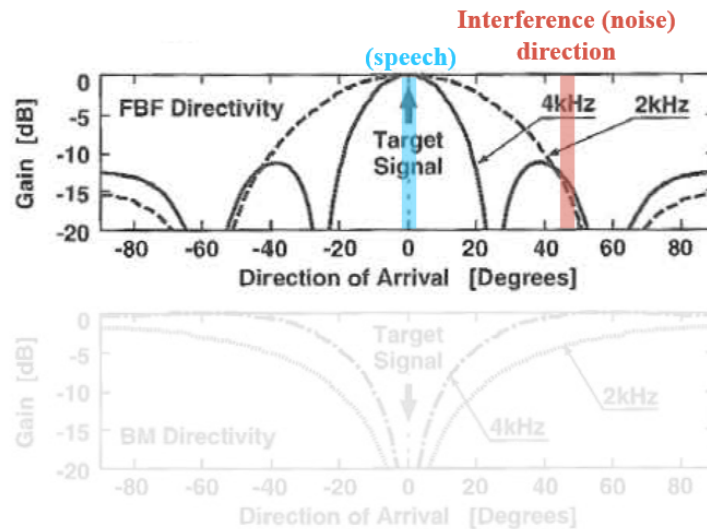


Fig. 5.3. Directivity pattern of a fixed beamformer (FBF) and a blocking matrix (BM).

EX1005, 90, Fig. 5.3 (annotated)

The FBF of the top branch (*first virtual microphone*) has *a first response to* the “*target signal*” (*speech*), at 0°, and a *first response to* interference (*noise*), at 45°. (EX1003, ¶99.)

⁶ The '080 Patent defines “noise” as “unwanted environmental acoustic noise” (EX1001, 4:29-30), which includes Brandstein’s reference to environmental acoustic noise as “interference,” described as air-conditioner noise or white noise. (EX1005, 101, 103, Fig. 5.13; EX1003 ¶97.)

Brandstein’s first virtual microphone’s *responses* to speech and noise are *linear* because the responses are represented by a linear system. (EX1003, ¶100.) For example, Brandstein’s Fig. 5.1 depicts the FBF as applying “fixed filter[s]” that are summed together, which are *linear* components. (*Id.*; EX1005, 95, Figs. 5.1, 5.4.)

Therefore, the Brandstein’s *first virtual microphone* has a *first linear response to speech* and a *first linear response to noise*. (EX1003, ¶100.)

- e. [7d] **“the first linear response to speech being substantially similar across a plurality of frequencies for a speech source located within a predetermined angle relative to an axis of the microphone array and devoid of a null,”**

Brandstein discloses or renders this limitation obvious. (*Id.*, ¶¶101-108.)

As explained in Section VII.A.1-2, a POSITA would have been motivated to modify the basic GJBF to form the GJBF-CNA, such that the directivity patterns relating to the basic GJBF and its components would have also applied to the GJBF-CNA. (*Id.*, ¶¶102-103.) Brandstein’s Fig. 5.10 plots “the sensitivities to the frequency component of the target signal” (*first linear response to speech across a plurality of frequencies*) at different frequencies for the modified GJBF:

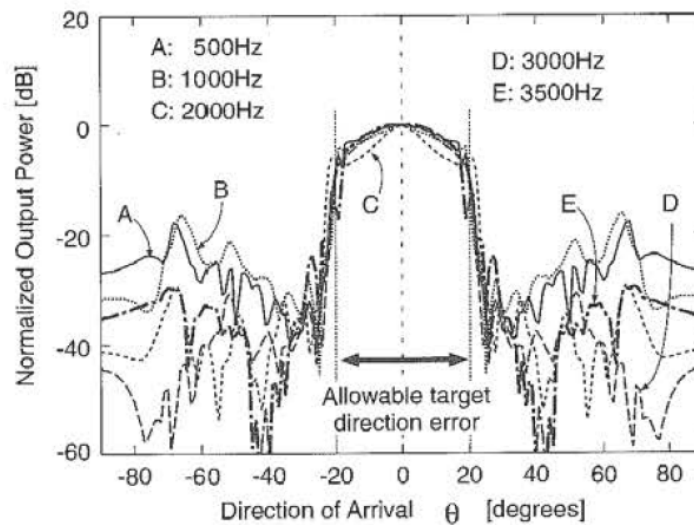


Fig. 5.10. Sensitivities after convergence as a function of DOA at different frequencies.

EX1005, 100, Fig. 5.10

(EX1003, ¶104.)

Brandstein explains that the “[f]requency dependency of the GJBF-CNA is small” and that the sensitivity in the region around the target direction (*i.e.*, direction of *speech*) “is constant.” (EX1005, 99-100, EX1003, ¶105.) Fig. 5.10 depicts the output $y(k)$ of Fig. 5.8, as it is described in relation to “the GJBF-CNA.” (EX1005, 100, EX1003, ¶105.) Moreover, because the GJBF bottom branch output in Fig. 5.8 has a null in the speech direction (because it has “a wider null for the target [speech] with sharper edges”), the POSITA would have further understood that the Fig. 5.10 plot in the target *speech* direction would have also been the top branch’s response in the same *speech* direction, for the different *frequencies* shown. (EX1003, ¶105.)

For these reasons, the top branch’s response to *speech* of the GJBF-CNA (*first response*) is constant (which meets *substantially similar*) *across a plurality frequencies*. (*Id.*)

Brandstein explains that in Fig. 5.10, the sensitivity is “constant” *within* the “high-sensitivity regions” across different frequencies for the range of angles - $20^\circ < \theta < 20^\circ$, where θ is *an angle relative to an axis of the microphone array* as shown in Fig. 5.1. (*Id.*, ¶106; EX1005, 100.) In Fig. 5.10, the “maximum allowable target-direction error” was 20° (EX1005, 100), which the POSITA would have understood refers to a pre-set threshold, or *a predetermined angle* defining a range (-20 to 20°) *within which a speech source is located*. (EX1003, ¶106.)

There is also no *null* in the target (*speech*) direction in Fig. 5.10 because there is no zero or minima in the FBF’s directivity pattern (*first linear response*) in that direction. (EX1001, 4:31-32; EX1003, ¶107.) Therefore Brandstein discloses a *first linear response to speech* that is *devoid of a null*. (EX1003, ¶107.)

Moreover, a POSITA would have understood that the top branch’s directivity pattern (*first linear response to speech*) is *devoid of a null* in all directions for certain frequencies. (*Id.*, ¶108.) The top plot in Fig. 5.3 depicts the main lobe (with the largest maximum) of the directivity pattern centered around 0° , the *speech* direction, and the main lobe width varies for different frequencies (as shown, the width doubled when frequency is halved from 4kHz to 2kHz). The POSITA would have

understood for certain low frequencies (*e.g.*, below 1kHz), the main lobe’s width would span an entire 360°, such that the *response is devoid of a null* in any direction.

(EX1003, ¶108).

- f. [7e] “wherein the second virtual microphone has a second linear response to speech that has a single null oriented in a direction toward a source of the speech and a second linear response to noise,”

Brandstein discloses this limitation for both mappings of *second virtual microphone* (whether the BM alone or the entire bottom branch). (*Id.*, ¶¶109-116.)

For the second mapping, as Brandstein depicts in Fig. 5.3, the BM (*second virtual microphone*) has *a second linear response to speech*, which is in the direction of a *speech* “target signal” (*Id.*, ¶110):

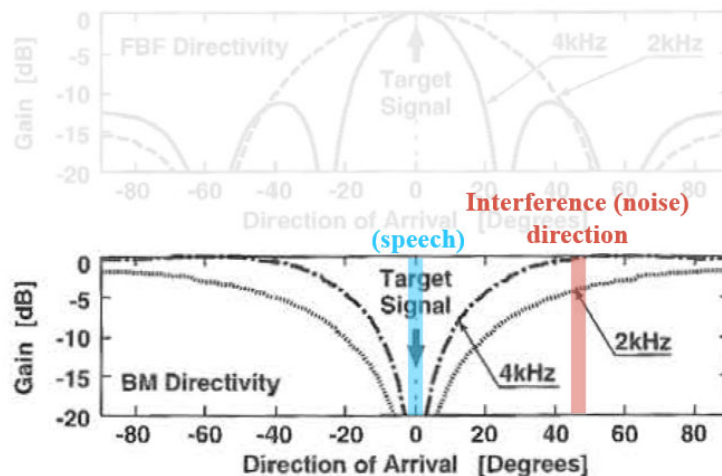


Fig. 5.3. Directivity pattern of a fixed beamformer (FBF) and a blocking matrix (BM).

EX1005, 90, Fig. 5.3 (annotated)

For the same reasons as discussed above in Section VII.A.3.d ([7c]), the BM (*second virtual microphone*) also has *a second response to noise* at the 45 degree interference direction, so that the BM response has *a single null oriented in the direction toward the target signal (a source of speech)*. (EX1003, ¶111.) Moreover, while Brandstein’s Fig. 5.3 does not expressly show the response in all directions (360°), the POSITA would have known that the response of the BM has a single null oriented in a direction toward a source of the speech for certain speech directions, such as if the speech direction is parallel to the axis of the microphone array. (EX1003, ¶¶112-114 (citing EX1023, Fig. 4a, depicting a polar sensitivity plot having a single null in all directions for a delay-and-subtract structure like that of the GJBF’s BM, where the sound source is aligned with the microphone array).)

The second virtual microphone’s responses to *speech* and *noise* are *linear* because the BM is represented by difference modules, adaptive filters, and a summer, all *linear* components. (*Supra* Section VII.A.3.d ([7c]); EX1003, ¶115.)

For the first mapping, while the Fig. 3 bottom plot is of the BM, a POSITA would have understood that a directivity pattern with the same overall characteristics would result at the bottom branch’s output, after filtering and summing. (EX1003, ¶116.) This is because the bottom branch’s noise only signal would have also had a null in the speech direction and higher responses elsewhere, as it has a “wider null... with sharper edges.” (*Id.*) Moreover, the components of the bottom branch are

linear for the same reason as stated for the BM alone, so the *responses to speech* and *noise* are *linear*. Thus, just like the BM output, the bottom branch’s output (*second linear response*) would have also had *a single null oriented in a direction toward a source of the speech* as well as *a second linear response to noise*. (*Id.*)

Therefore, Brandstein’s teachings with respect to the *second linear response to speech* and *noise* are applicable under both first and second mappings. (*Id.*)⁷

- g. [7f] “wherein the second linear response to noise is substantially similar to the first linear response to noise,” and [7h] “the second linear response to speech is substantially dissimilar to the first linear response to speech,”**

Brandstein discloses or renders obvious these limitations. (EX1003, ¶¶117-122.)

⁷ Because the overall characteristics of the directivity patterns for the BM output (first mapping for *second virtual microphone*) and the bottom branch’s output (second mapping for *second virtual microphone*) are the same, a mapping for the *second virtual microphone* or *the second linear response to speech* or *noise* to the BM output would also apply to the bottom branch’s output, and vice versa. (EX1003, ¶¶111-116.) Accordingly, Petitioner does not repeat an explanation for both mappings for every claim limitation, and for such claim limitations, Petitioner’s explanation applies to both mappings.

For both mappings, Brandstein’s virtual microphones have very different (meeting *substantially dissimilar*) responses to the target signal (*speech*). The top branch (*first virtual microphone*) forms a beam capturing the target, while the BM or bottom branch (both mappings for *second virtual microphone*) blocks the target. (EX1005, 88; EX1003, ¶118)

The two virtual microphones also have essentially the same (meeting *substantially similar*) responses to noise, so that the noise responses cancel out in the GJBF’s final subtraction. (EX1003, ¶119.)

The very different *speech* responses and nearly identical *noise* responses are evident from Figs. 5.2 and 5.3 (*Id.*, ¶119):

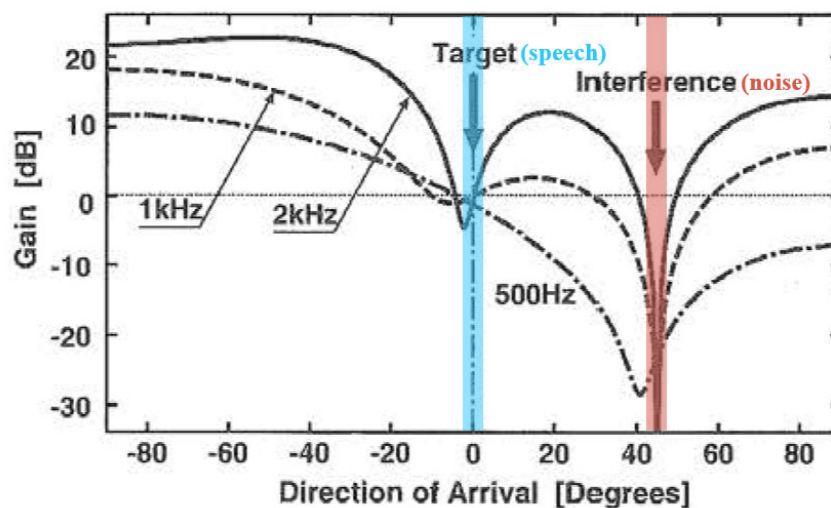


Fig. 5.2. Example directivity pattern of the Griffiths-Jim beamformer.

EX1005, 89, Fig. 5.2 (annotated)

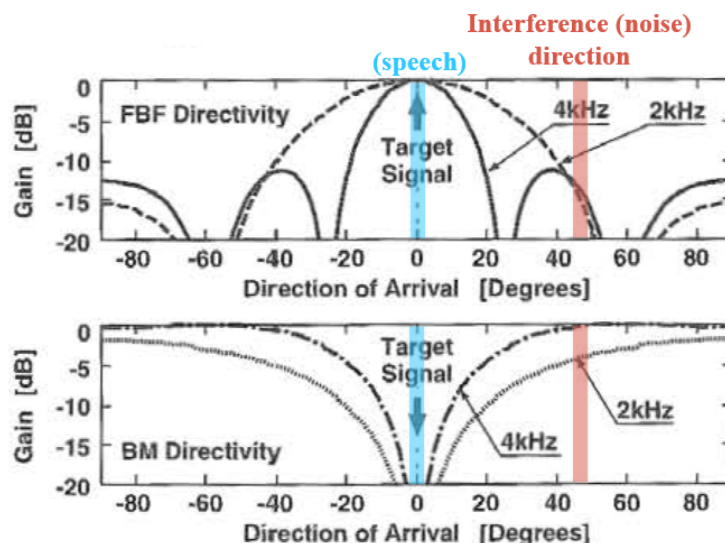


Fig. 5.3. Directivity pattern of a fixed beamformer (FBF) and a blocking matrix (BM).

EX1005, 90, Fig. 5.3 (annotated)

As highlighted in blue, in the *speech* direction, the GJBF achieves “almost constant gains close to 0 dB.” (*Id.*, 90; EX1003, ¶120.) This means that subtracting the bottom branch’s noise only output from the top branch’s speech and noise output produces the target signal in the *speech* direction, confirming the two virtual microphones have very different responses to the target signal, which is sufficient to meet the second linear response to speech is substantially dissimilar to the first linear response to speech. (EX1003, ¶120.) This is also confirmed in Fig. 5.3, showing that the FBF’s response has a maximum in the *speech* direction, which is very different from the BM’s null in the same direction.

In contrast, as highlighted in red in Fig. 5.2, “in the direction of the interference, a deep null is formed.” (EX1005, 90; EX1003, ¶121.) The deep null

reflects that subtracting the bottom branch from the top branch cancels the interference, confirming that the virtual microphones’ responses to the interference are essentially the same, which is sufficient to meet *the second linear response to noise is substantially similar to the first linear response to noise*. (EX1003, ¶121.)

Thus, Brandstein’s teachings of the *second linear response to speech* and *noise* are applicable under both first and second mappings because the directivity pattern of the bottom branch would have had the same overall characteristics as the BM shown in Fig. 5.3’s bottom plot. (*Id.*, ¶122.)

h. [7g] “one or both of the first linear response to noise and the second linear response to noise being nonzero in a direction toward a source of noise, and”

Brandstein discloses or renders obvious this limitation. (*Id.*, ¶¶123-124.)

As depicted in Fig. 5.3, the only null in the directivity pattern of the *second virtual microphone* (in either mapping) is in the speech direction. In all other directions, including *toward a source of noise*, the response of the BM or bottom branch (*second linear response*) is nonzero. (*Id.*, ¶124.)

i. [7i] “wherein the speech is human speech; and”

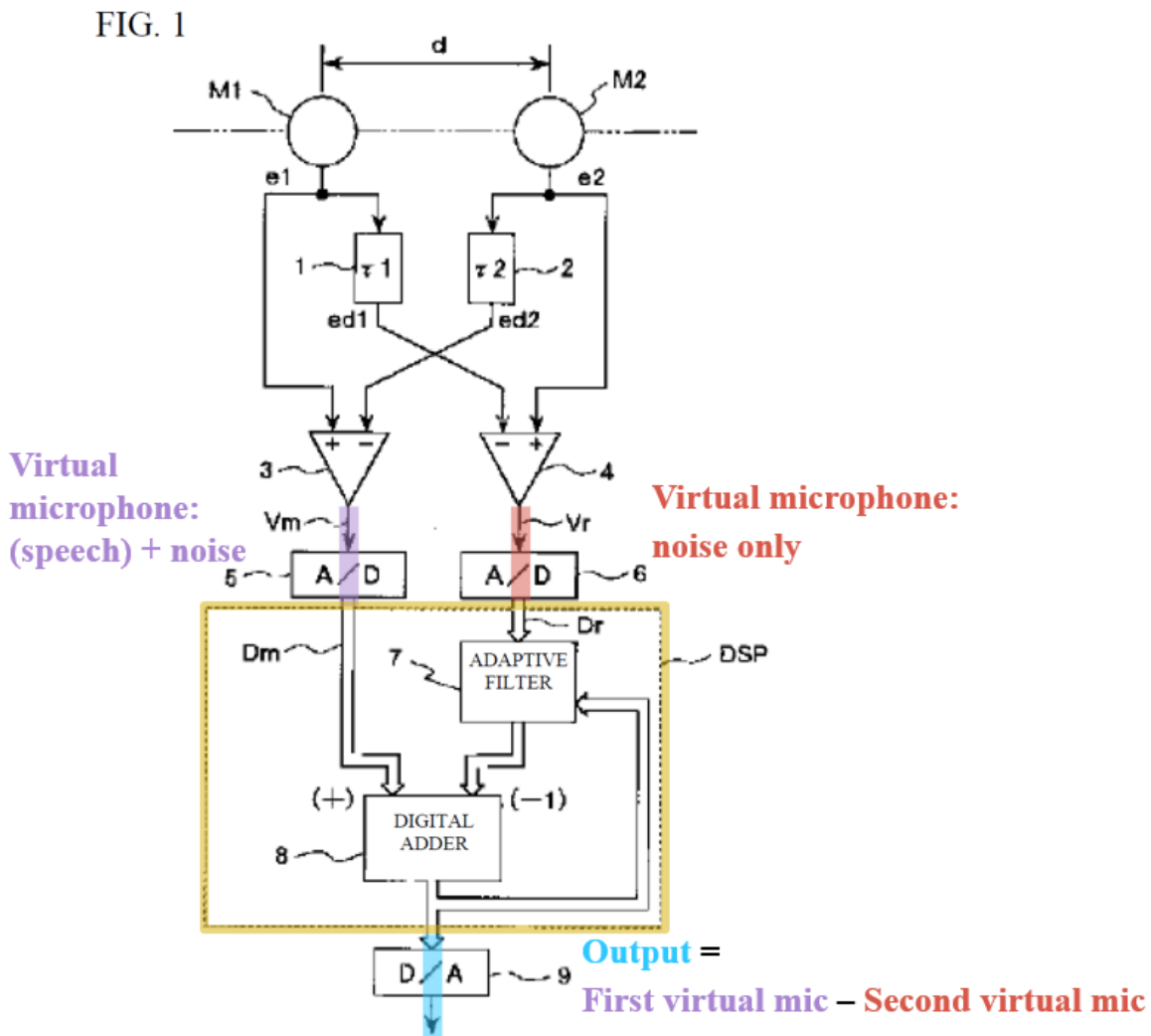
Brandstein discloses this limitation. (EX1003, ¶¶125-126.)

In Brandstein, the target *speech* signal is *human speech*, like “an English male speech signal.” (*Id.*; EX1005, 101.)

- j. [7j] “an adaptive noise removal application coupled to the first and second virtual microphones and generating denoised output signals”

Brandstein and Ikeda render obvious this limitation. (EX1003, ¶¶127-134.)

Ikeda’s **adaptive noise removal system** includes adaptive filter 7 and digital adder 8, is implemented using a DSP, and is therefore an *application*. (EX1008, ¶¶[0016]-[0017], [0027]-[0030], Fig. 1; EX1003, ¶128.)



EX1008, Fig. 1 (annotated).

Ikeda's *adaptive noise removal application* receives main signal Vm (analogous to Brandstein's *first virtual microphone*) and reference signal Vr (analogous to *second virtual microphone*),⁸ and is therefore *coupled to the first and second virtual microphones* via A/D converters 5 and 6. (*Id.*, ¶¶[0027]-[0029], Fig. 1; EX1001, 32:25-32; EX1003, ¶129.)

Ikeda's *adaptive noise removal application* generates denoised output signals because Ikeda discloses that “the noise component contained in the main signal Vm is cancelled by the noise component contained in the reference signal Vr due to subtraction by digital adder 8.” (EX1008, ¶¶[0028]-[0029]; EX1003, ¶130.)

Under both mappings, the *first virtual microphone* and the *second virtual microphone* are *coupled to* Ikeda's DSP (*adaptive noise removal application*). (EX1003, ¶131.)

For the first mapping, the outputs of the top and bottom branches are *coupled to* Ikeda's DSP:

⁸ As discussed above in Sections VII.A.1 and VII.A.3.c ([7b]), the second virtual microphone signal is the signal output from the bottom branch (first mapping) or the BM (second mapping). The teachings of Ikeda are equally applicable to either mapping. (EX1003, ¶129.)

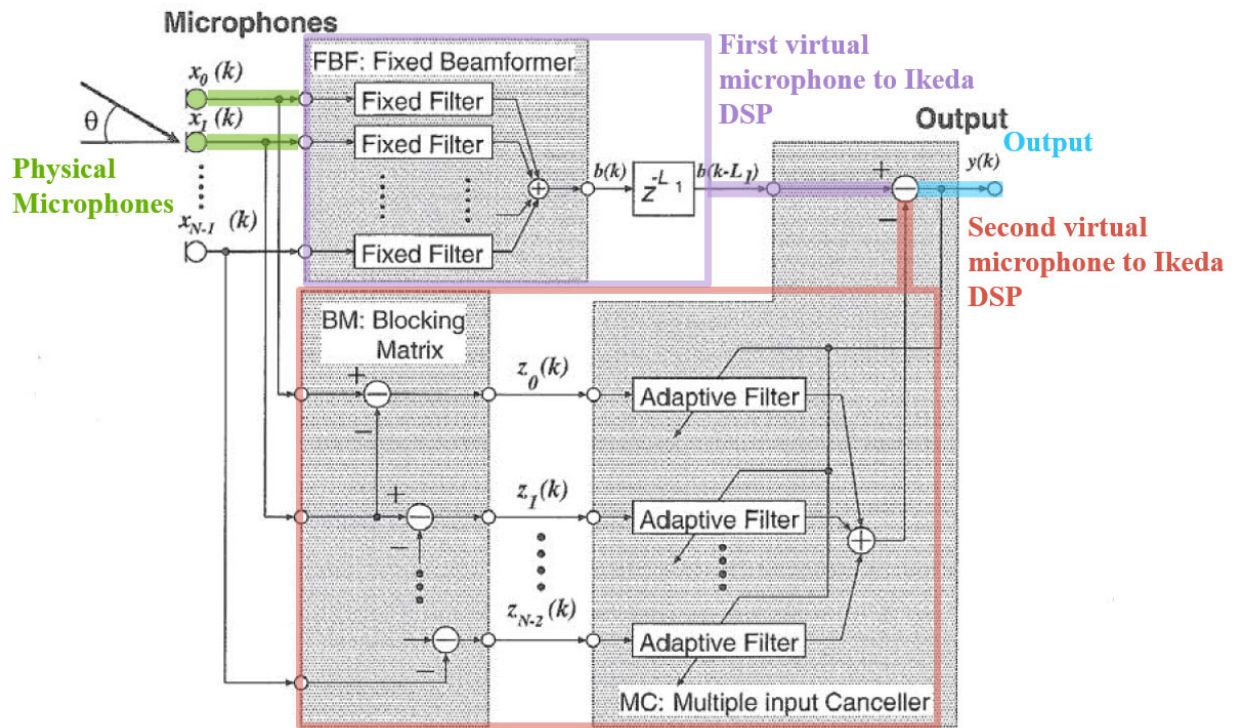


Fig. 5.1. Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceller (MC), and a blocking matrix (BM).

EX1005, 89, Fig. 5.1 (annotated)

(EX1003, ¶132.)

For the second mapping, the outputs of the top branch and BM are *coupled to* Ikeda’s DSP, operating in a similar manner as Brandstein’s MC of a two-microphone array (EX1003, ¶133):

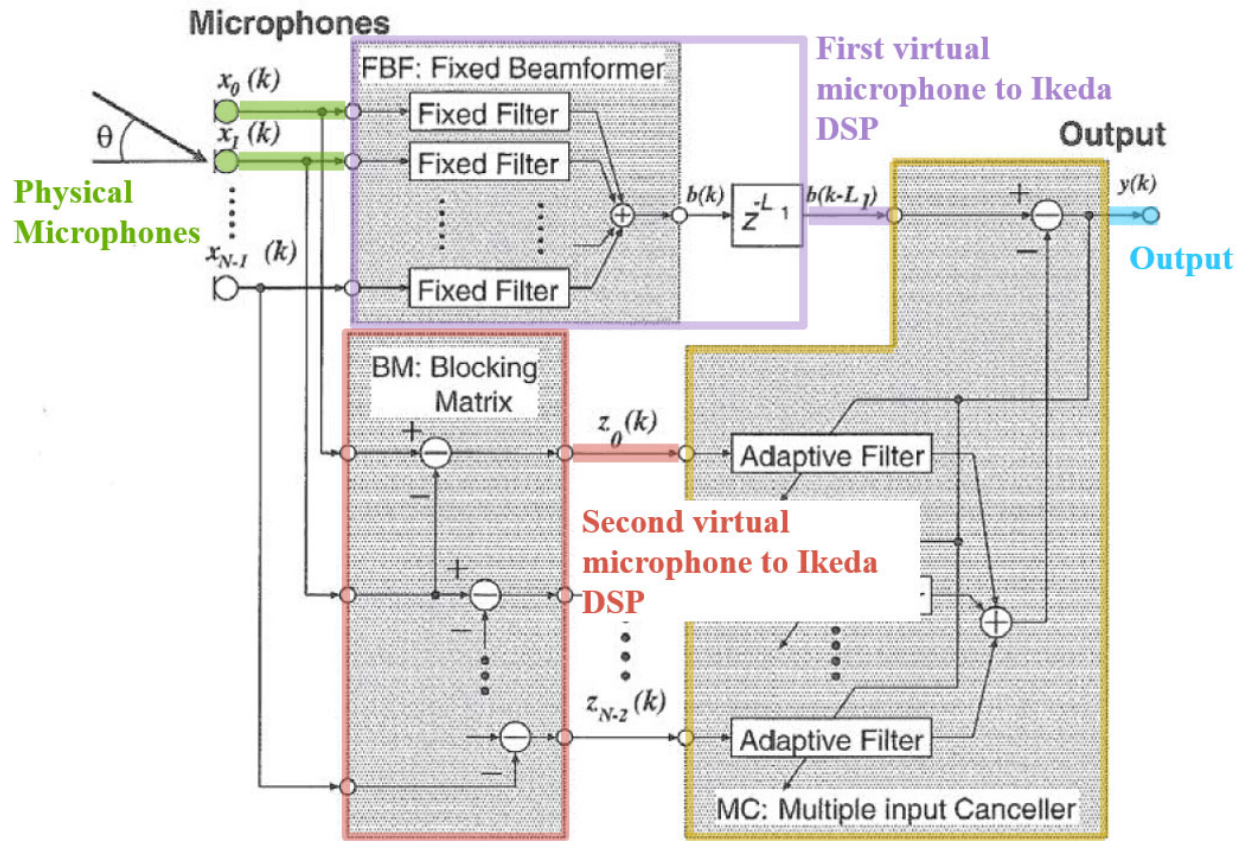


Fig. 5.1. Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceler (MC), and a blocking matrix (BM).

EX1005, 89, Fig. 5.1 (annotated)

In both mappings, the output of the top branch (*the first virtual microphone*) is input into Ikeda's DSP (*coupled to the adaptive noise removal application*). Ikeda's DSP subtracts the output of the adaptive filter 7 from the main signal V_m (which is analogous to the top branch's output and containing speech and noise) to produce an output that is fed back into the adaptive filter 7. (EX1003, ¶134.)

k. [7k] “by forming a plurality of combinations of signals output from the first virtual microphone and the second virtual microphone,”

Brandstein and Ikeda render obvious this limitation. (*Id.*, ¶¶135-136.)

Under both mappings, Ikeda’s adaptive noise removal application forms a *plurality of combinations of signals output from the first virtual microphone and the second virtual microphone* because Vm and Vr are continuously converted to digital signals and input to the DSP’s adaptive noise removal system so that, over time, they form multiple combinations of signals output from the first and second virtual microphones. (EX1008, ¶¶[0027]- [0029]; EX1003, ¶136; *see also* EX1001, 32:61-63.) This is consistent with the ’080 Patent’s description that an adaptive noise removal system “combin[es] the two microphone signals (*e.g.*, Mic1, Mic2) by filtering and summing in the time domain.” (EX1001, 7:4-16; EX1003, ¶136; *see also* EX1001, 8:50-9:3, 9:22-38.)

l. [7l] “by filtering and summing the plurality of combinations of signals in the time domain, and”

Brandstein and Ikeda render obvious this limitation. (EX1003, ¶¶137-138.)

Under both mappings, Ikeda’s adaptive noise removal system *filters and sums the combinations of signals in the time domain* because digital versions of Vr are “supplied to an adaptive filter” (*filtering*) and the output of adaptive filter 7 and digital versions of Vm are differenced at digital adder 8 (*summing*). (EX1008, ¶¶[0027]-[0029]; EX1003, ¶138.) This is consistent with the ’080 Patent’s

description that an adaptive noise removal system “combin[es] the two microphone signals (*e.g.*, Mic1, Mic2) by filtering and summing in the time domain.” (EX1001, 7:4-16; EX1003, ¶138; *see also* EX1001, 8:50-9:3, 9:22-38.) A POSITA would have understood that adaptive filter 7 filters, and digital adder 8 adds, in the time domain because Ikeda does not include components for converting signals from time-to-frequency domain or vice versa, and a signal picked up by a microphone is in the time domain if not processed and converted to frequency domain. (EX1008, ¶¶[0027]-[0029]; EX1010, [0042]-[0043], [0058]-[0062]; EX1003, ¶138.) Additionally, the POSITA would have found it straightforward and known to implement Ikeda’s adaptive filter and digital adder to process signals in the time domain. (*Id.*)

m. [7m] “by a varying linear transfer function between the plurality of combinations of signals,”

Brandstein and Ikeda render obvious this limitation. (*Id.*, ¶¶139-141.)

Under both mappings, Ikeda’s adaptive noise removal system *generat[es] denoised output signals... by a varying transfer function between the plurality of combinations of signals*. (*Id.*, ¶140.) Ikeda’s adaptive filter 7’s characteristics “are varied in real time” by “a tap coefficient updating operation for updating the tap coefficient based on an error signal obtained as the difference between the filter output and a signal from the target system to return the filter output,” which is successively repeated to rewrite the tap coefficient. (EX1008, ¶¶[0027]-[0029].)

This “error signal is defined as the output of the digital adder 8,” where the *plurality of combinations* of signals are *summed*. (*Id.*, ¶[0028].) Ikeda performs this tap coefficient updating operation when “the microphone units M1 and M2 do not pick up the target sound,” which is when the output of digital adder 8 is zero (because there is no speech). (*Id.*) By updating the tap coefficient over time to minimize this error signal, Ikeda’s adaptive filter calculates and applies a *varying transfer function between the plurality of combinations of signals*, resulting in the output of adaptive filter 7 equaling the digital signal from main signal Vm when there is no speech. (*Id.*; EX1003, ¶140.) This is consistent with the ’080 Patent’s description that “[t]he adaptive filter generally uses the signal received from a first microphone... to remove noise from the speech received from at least one other microphone..., which relies on a slowly varying linear transfer function between the two microphones for sources of noise.” (EX1001, 7:4-16; EX1003, ¶140; *see also* EX1001, 5:19-44, 6:26-29, 8:50-9:3, 9:22-38.)

Ikeda’s *transfer function* is *linear* because adaptive filter 7 performs linear operations: “multiplying the input signal Dr by a tap coefficient and adding the value in multiple tap units.” (EX1008, ¶¶[0027]-[0029]; EX1003, ¶141; EX1013, 16-17, Fig. 2.2; EX1014, 366-367.) A POSITA would also have known to implement adaptive filter 7 as *linear*, as such filters were well-known for adaptive noise cancellation, and would have been an obvious design choice. (EX1015, 4:28-5:15,

Fig. 2; EX1003, ¶141.)

- n. [7n] “wherein the denoised output signals include less acoustic noise than acoustic signals received at the first and second physical microphones.”

Brandstein and Ikeda each discloses this limitation. (EX1003, ¶¶142-144.)

Both references remove or cancel acoustic noise received from physical microphones in a microphone array. (*Id.*, ¶143.) Applications of Brandstein’s beamforming include speech enhancement for “remov[ing] undesirable signals such as noise and reverberation.” (EX1005, 87). As seen in Fig. 5.2 and explained above in Section VII.A.3.g ([7f]), the denoised output signal *includes less acoustic noise* because the deep null in the *noise* direction means that noise is suppressed compared to the physical microphone signals. (EX1005, 89-90 (“[T]arget signal extraction and interference suppression are simultaneously achieved.”); EX1003, ¶143.)

Ikeda also discloses this limitation because, unlike the microphone signals, the denoised output signals have had “the noise component contained in the main signal V_m ” canceled and “primarily consist[] of the target sound component.” (EX1008, ¶¶[0001], [0011]-[0012], [0029], Abstract; EX1003, ¶144.)

4. **Claim 8: “The system of claim 7 and further comprising: a microphone array, the first and second physical microphones positioned m [*sic*] the microphone array.”**

Brandstein discloses this limitation. (EX1003, ¶¶145-146.)

As discussed above in Section VII.A.3.b ([7a]), Brandstein’s Fig. 5.1 shows a *microphone array* that includes *first and second physical microphones* that generate first and second microphone signals (e.g., x_0 and x_1) *positioned in the array*. (EX1005, 89, Fig. 5.1; EX1003, ¶146.)

5. **Claim 9: “The system of claim 7, wherein the single null is a region of the second linear response to speech having a measured response level that is lower than the measured response level of any other region of the second linear response to speech.”**

Brandstein discloses this limitation. (EX1003, ¶¶147-149.)

The *second virtual microphone* (BM or bottom branch of the GJBF) has a *null* or minimum in the *speech* direction:

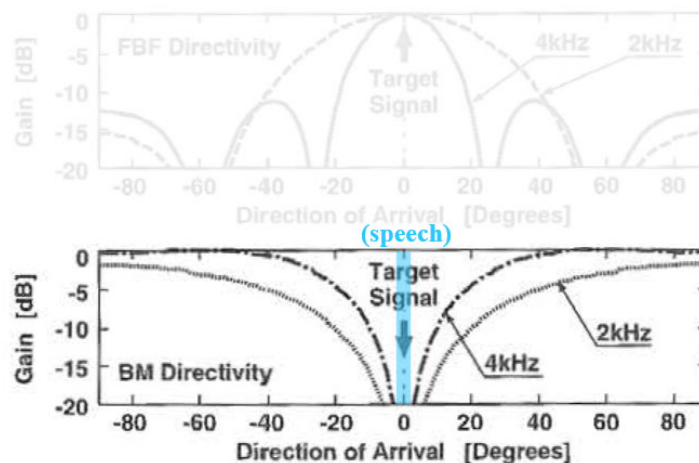


Fig. 5.3. Directivity pattern of a fixed beamformer (FBF) and a blocking matrix (BM).

EX1005, 90, Fig. 5.3 (annotated)

Therefore, the *null* is a region of the *second* virtual microphone’s *linear response to speech having a measured response level that is lower than a measured response level of any other region of the second linear response to speech*. (EX1003, ¶¶147-149.)

6. Claim 11: “The system of claim 7 and further comprising: a communications channel coupled with the processing component and including one or more of a wireless channel, a wired channel, and a hybrid wireless/wired channel.”

Brandstein in combination with Ikeda renders this limitation obvious.⁹ (*Id.*, ¶¶150-153.)

Ikeda discloses a *processing component* as a DSP including an “adaptive filter 7.” (EX1008, ¶¶[0016]-[0017], [0030]; EX1003, ¶151.)

Brandstein’s beamforming applications include microphone arrays for speech enhancement in “hands free telephones.” (EX1005, 87.) A POSITA would have understood that there is a *communications channel* between such GJBF microphone arrays and the hands free telephone. (EX1003, ¶152.) As the POSITA would have readily appreciated, a *communications channel* relates to the flow of information (signal) from one point to another, and Ikeda’s DSP is along a *communication*

⁹ “[T]he processing component” element of claim 11 lacks antecedent basis. Petitioner assumes that claim 11 requires “a processing component.”

channel, as it receives inputs from both *virtual microphones* and provides an output corresponding to the difference between the speech plus noise signal and the adaptively filtered noise signal. (*Id.*) A POSITA would therefore have understood that Ikeda’s DSP represents part of the *communications channel*, which is coupled with the processing component in the Brandstein-Ikeda system. (*Id.*)

A POSITA would have further known that the *communications channel* includes one or more of a wireless channel, a wired channel, and a hybrid wireless/wired channel, because all channels are wireless, wired, or both. (*Id.*, ¶153.) Moreover, as Ikeda’s DSP is wired and Brandstein’s “hands free telephone” includes a wireless component, the POSITA would have known to implement the *communication channel* to be *wireless, wired, or hybrid wireless/wired*. (*Id.*; EX1016, 2:58-64, Fig. 2; EX1017, 4:1-5:25, Figs. 3-4; EX1018, 3:20-25, 3:50-64, Claim 1, Fig. 4b; EX1019, ¶¶[0029]-[0030]; EX1020, ¶[0138].)

7. Claim 12: “The system of claim 11 and further comprising: a communication device wirelessly coupled with the wireless channel of the communications channel.”

Brandstein and Ikeda render this limitation obvious. (EX1003, ¶¶154-155.)

Brandstein’s “hands-free telephone” is a *communication device*. (EX1005, 87; EX1003, ¶155.) As discussed for claim 11, the *communications channel* is between the “microphone array” and the “hands-free telephone.” (EX1003, ¶155.) A POSITA would have understood that a hands-free telephone operates wirelessly and

that any hands-free device (*communication device*) is *wirelessly coupled with the wireless channel of the communications channel*. (*Id.*)

8. **Claim 13: “The system of claim 7, wherein the second microphone signal is multiplied by a ratio, wherein the ratio is a ratio of a third distance to a fourth distance, the third distance being between the first physical microphone and the speech source and the fourth distance being between the second physical microphone and the speech source.”**

Brandstein discloses or renders obvious this limitation. (*Id.*, ¶¶156-162.)

A POSITA would have known that for applications involving near-field targets, the GJBF requires time-alignment by applying delays, and for near-field beamforming, this is performed via a standard near-field propagation model. (EX1005, 25, Eqs. 2.28-30; EX1003, ¶157; *supra* Section II.E.) It would have been obvious to apply Brandstein’s Chapter 2 discussion of near-field time-alignment to its Chapter 5 GJBF description. (EX1003, ¶¶157-160.) Brandstein states it is straightforward to include near-field assumptions for time-alignment. (EX1005, 34-37; *see also id.* 290.) Brandstein therefore motivates using delays for near-field time-alignment in a GJBF. Moreover, in describing the GJBF’s BM, Brandstein states its “simplest structure... was described in the previous section,” which is Chapter 2. (EX1005, 88, 90; EX1003, ¶¶161-162.)

Combining the teachings of Brandstein’s Chapters 2 and 5 to arrive at this limitation is a simple addition of one known element (near-field time alignment for

near-field sources) to another known element (GJBF) to obtain predictable results (effective target-signal blocking in the lower branch). (EX1003, ¶¶161-162.) A POSITA would have had a reasonable expectation of success, and this limitation would therefore have been obvious. *KSR*, 550 U.S., 416-17.

Brandstein’s Chapter 2 discloses that time-aligning signals for near-field beamforming applies a standard near-field propagation model:

- Nearfield design, including attenuation of the desired signal [14], [22]

$$\mathbf{d}^T = [a_0 \exp(-j\omega\tau_0), a_1 \exp(-j\omega\tau_1), \dots, a_{N-1} \exp(-j\omega\tau_{N-1})], \quad (2.28)$$

$$a_i = \frac{\|\mathbf{q} - \mathbf{p}_{ref}\|}{\|\mathbf{q} - \mathbf{p}_i\|}, \quad (2.29)$$

$$\tau_i = \frac{\|\mathbf{q} - \mathbf{p}_{ref}\| - \|\mathbf{q} - \mathbf{p}_i\|}{c}, \quad (2.30)$$

where $\|\mathbf{q} - \mathbf{p}_{ref}\|$ and $\|\mathbf{q} - \mathbf{p}_i\|$ denote the distance between the vector location of the source $\mathbf{q}$ and a reference sensor $\mathbf{p}_{ref}$, or the sensor $\mathbf{p}_i$, respectively.

(EX1005, 25; EX1003, ¶161.). Brandstein’s Eq. 2.29 expresses a microphone attenuation factor a_i as a *ratio* between the distances from *speech source* $\mathbf{q}$ to *physical microphones* $\mathbf{p}_{ref}$ and $\mathbf{p}_i$, which are the claimed *third and fourth distances*. Brandstein’s a_i (*ratio*) is *multiplied by the i th (e.g., second) microphone signal* to achieve equal amplitudes across all channels. (EX1003, ¶161.)

Thus, a POSITA would have understood that Brandstein applies Eq. 2.28’s complex-exponential term to delay a signal in proportion to τ_i to time-align the

signals because: (1) the BM has a delay-and-sum (or delay-and-subtract) structure; (2) whose purpose is to time-align the sensor signals; (3) delay-and-sum beamforming is extended to near-field designs; and (4) the standard near-field design specifies a delay whose magnitude is proportional to the difference in propagation time of the target signal to one microphone relative to another (reference) microphone. (EX1003, ¶162.)

9. Independent Claim 14

Claim 14 is similar to independent claim 7, with a few exceptions which are mapped below. *See* analysis for cl. 7 in Section VII.A.3, which is incorporated herein. Specifically, the mappings of the following limitations apply to claim 14. (EX1003, ¶¶163-178.)

- [14pre] applies the mapping from [7pre] (Section VII.A.3.a);
- [14a] applies the mapping from [7a] (Section VII.A.3.b);
- [14b] applies the mapping from [7c] (Section VII.A.3.d);
- [14c] applies the mapping from [7d] (Section VII.A.3.e);
- [14d] applies the mapping from [7a] (Section VII.A.3.b);
- [14e] applies the mapping from [7b] (Section VII.A.3.c);
- [14f] applies the mapping from [7e] (Section VII.A.3.f);
- [14g] applies the mapping from [7f] (Section VII.A.3.g);
- [14h] applies the mapping from [7g] (Section VII.A.3.h);

- [14i] applies the mapping from [7h] (Section VII.A.3.g);
- [14j] applies the mapping from [7b] (Section VII.A.3.c); and
- [14m] applies the mappings from [7j]-[7n] (Sections VII.A.3.j-n).

(EX1003, ¶¶163-178.)

a. [14k] “wherein the first virtual microphone and the second virtual microphone are distinct virtual directional microphones; and”

Brandstein discloses this limitation for similar reasons as discussed in Section VII.A.3.c regarding [7b], because having different combinations forming the *first* and *second virtual microphones* means that the *first* and *second virtual microphones are distinct*. (EX1003, ¶¶174-175.) Specifically, the GJBF signals from the top branch (*first virtual microphone*) are *distinct* from those from the bottom branch or BM (*second virtual microphone* in first and second mappings) because the first and second virtual directional microphones are formed with different combinations of signals from the microphone array (*i.e.*, different combinations of $x_0(k)$ and $x_1(k)$). (EX1005, 89, Fig 5.1.) Both virtual microphones are *directional* because the directivity patterns of the top and bottom branches (and BM) are not constant in all directions, as shown in Fig. 5.3. (*Id.*, 90, Fig 5.3; EX1003, ¶175.)

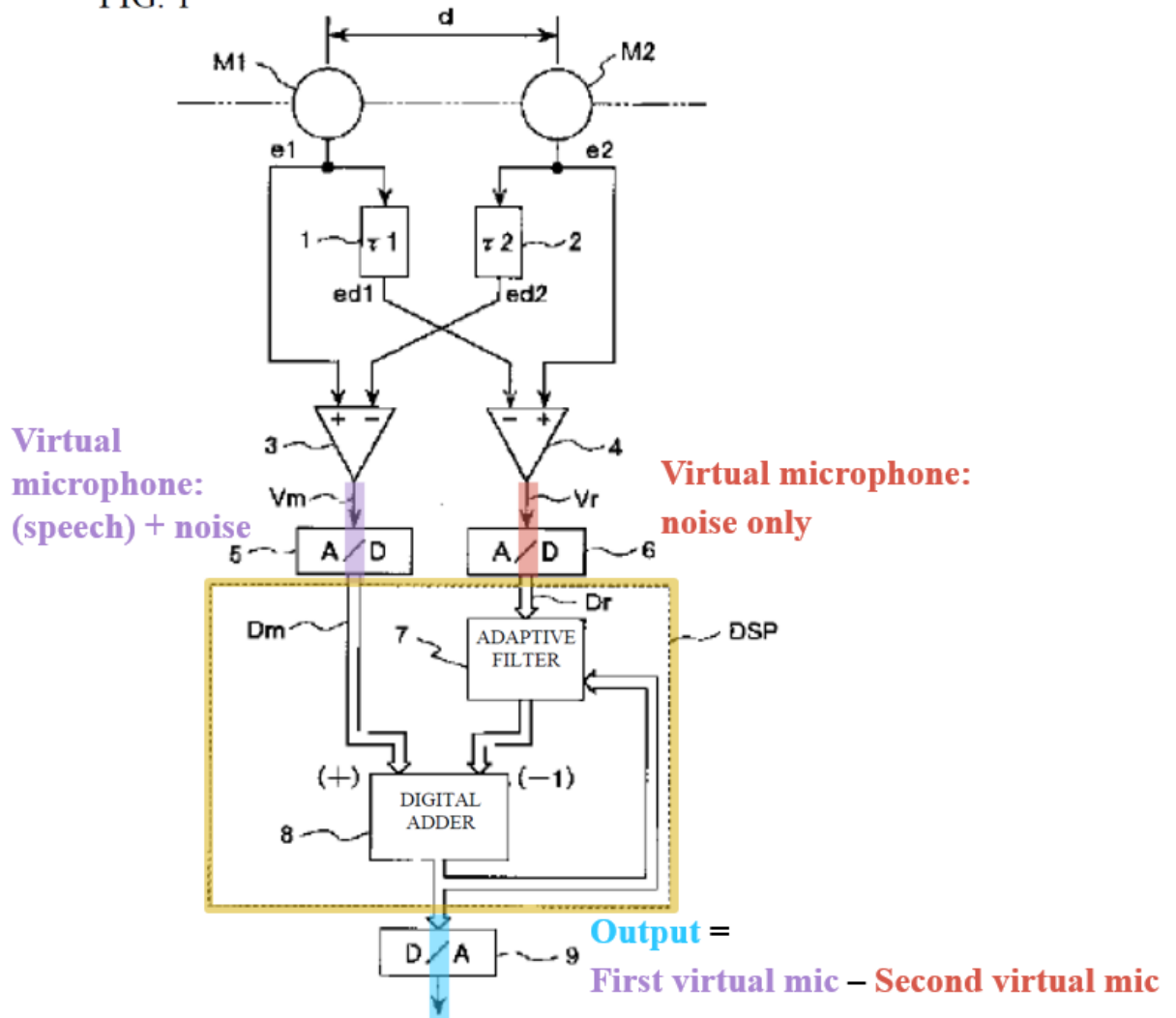
b. [14l] “a processing component coupled to the first and second virtual microphones, the processing component including an adaptive noise removal application receiving acoustic signals from the first virtual microphone and the second virtual

microphone, filtering and summing the acoustic signals in the time domain, applying a varying linear transfer function between the acoustic signals, and generating an output signal,”

Ikeda discloses this limitation for similar reasons as discussed in Sections VII.A.3.j-VII.A.3.m regarding [7j]-[7m]. (EX1003, ¶¶176-177.)

Regarding the *processing component*, Ikeda discloses that a DSP as shown in Fig. 1 (annotated below) is *a processing component* including an adaptive noise removal application. (EX1008, ¶¶[0016]-[0017], [0027]-[0030]; EX1003, ¶177). In the combination, the DSP is *coupled* to the *first and second virtual microphones*. (EX1008, ¶[0027], Fig. 1; EX1001, 32:25-32; EX1003, ¶177.)

FIG. 1



EX1008, Fig. 1 (annotated).

10. **Claim 16:** “The system of claim 14 and further comprising: a communications channel coupled with the processing component and including one or more of a wireless channel, a wired channel, and a hybrid wireless/wired channel.”

Brandstein and Ikeda render this limitation obvious for the same reasons discussed in Section VII.A.6 regarding claim 11. (EX1003, ¶178.)

- 11. Claim 17: “The system of claim 16 and further comprising: a communication device wirelessly coupled with the wireless channel of the communications channel.”**

Brandstein and Ikeda render this limitation obvious for the same reasons discussed in Section VII.A.7 regarding claim 12. (*Id.*, ¶180.)

- 12. Claim 18: “The system of claim 14, wherein the acoustic signals from the first virtual microphone, the second virtual microphone or both are delayed.”**

Brandstein discloses this limitation. (*Id.*, ¶¶181-182.)

Brandstein discloses that the signal in the top branch, after the FBF, is *delayed*. (*Id.*, ¶182.) Brandstein’s FBF output $b(k)$ is *delayed* through the block Z^{-L_1} to form the “delayed output signal $b(k-L_1)$ ” (EX1005, 88), highlighted below in yellow:

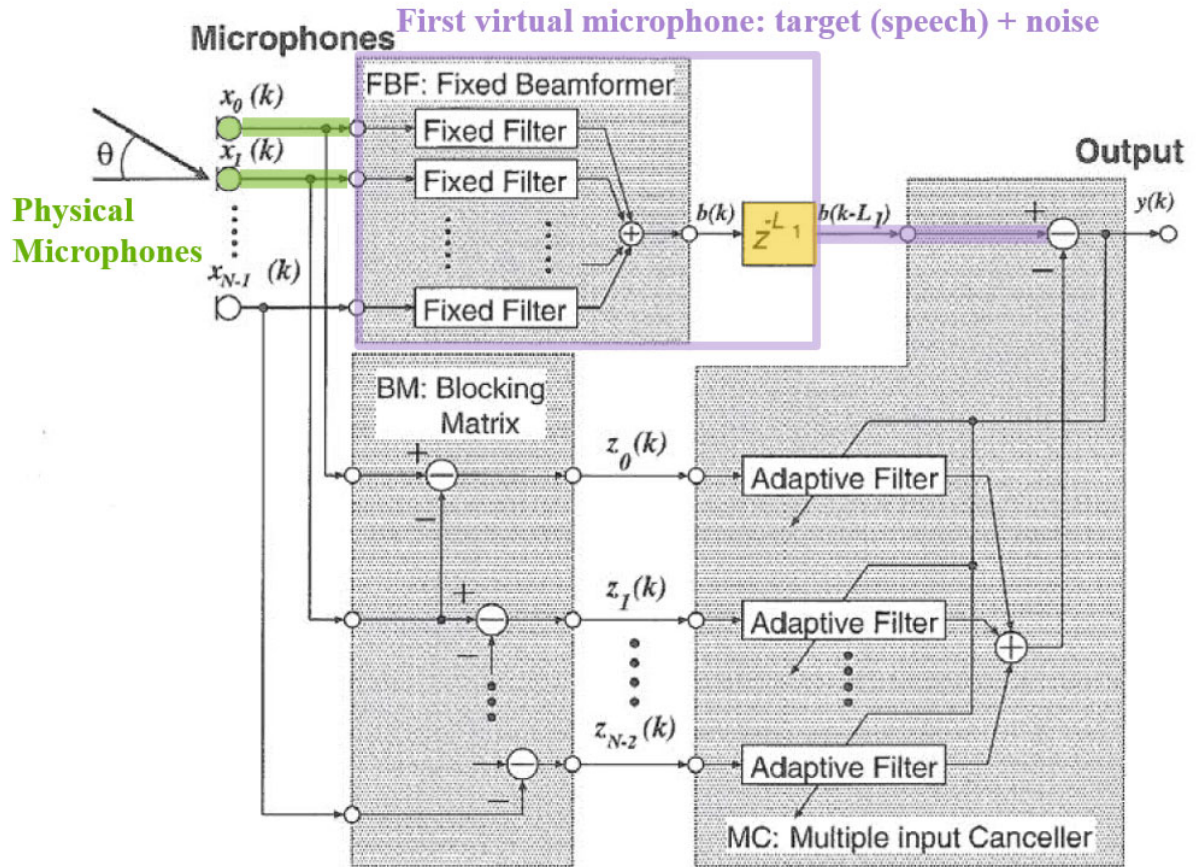


Fig. 5.1. Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceller (MC), and a blocking matrix (BM).

EX1005, 89, Fig. 5.1 (annotated).

Therefore, at least the *acoustic signals* from the top branch (*first virtual microphone*) are *delayed*. (EX1003, ¶182.)

13. **Claim 19:** “The system of claim 18, wherein the delay is raised to a power that is proportional to a time difference between arrival of the speech at the first virtual microphone and arrival of the speech at the second virtual microphone.”

Brandstein renders obvious this limitation. (*Id.*, ¶¶183-185.)

The '080 Patent defines:

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

where “[t]he distances d_1 and d_2 are the distance from O_1 and O_2 to the speech source..., and γ is their difference divided by c , the speed of sound, and multiplied by the sampling frequency f_s .” (EX1001, 9:64-67.) The '080 Patent refers to γ as “the time difference between arrival of speech at V_1 compared to V_2 .” (*Id.*, 13:66-67.) Thus, according to the '080 Patent, the *time difference* between arrival of speech at the virtual microphones is calculated from the difference between distances from the physical microphones to the speech source. (EX1003, ¶183.)

As discussed above for claim 13, Brandstein renders obvious extending the standard GJBF to a near-field design, by time-aligning the microphone signals fed to the BM by applying delay term $e^{-j\omega\tau_i}$, whose exponent (*power*) is *proportional* to τ_i . (*Supra* Section VII.A.8; EX1003, ¶184.) Brandstein defines τ_i as:

$$\tau_i = \frac{\|q - p_{ref}\| - \|q - p_i\|}{c}, \quad (2.30)$$

where $\|q - p_{ref}\|$ and $\|q - p_i\|$ denote the distance between the vector location of the source q and a reference sensor p_{ref} , or the sensor p_i , respectively.

(EX1005, 25.)

Brandstein’s τ_i results from dividing the *difference* of the distance between the source (*speech*) q and a reference sensor (*second microphone*)¹⁰ and the distance between the source and the *i*th sensor (*first microphone*) by the speed of sound. (EX1003, ¶185.) Brandstein’s τ_i is thus *a time difference between arrival of the speech at the first virtual microphone and arrival of the speech at the second virtual microphone*. (*Id.*)

14. **Claim 20: “The system of claim 19, wherein the power is proportional to a sampling frequency multiplied by a quantity equal to a third distance subtracted from a fourth distance, the third distance being between a first physical microphone and the speech source, the fourth distance being between a second physical microphone and the speech source, and the first and second physical microphones are positioned in the microphone array.”**

Brandstein discloses this limitation. (*Id.*, ¶¶186-191.)

¹⁰ For purposes of claim 19, Petitioner maps Brandstein’s *i*th and reference sensors to the *first* and *second microphones*, which differs from Petitioner’s claim 13 mappings. The POSITA would have understood that both claims are met by Brandstein’s teachings in different ways. (EX1003, ¶185.)

Brandstein's *first and second physical microphones are positioned in the microphone array* for the same reasons discussed in Section VII.A.4 regarding claim 8. (*Id.*, ¶187.)

As explained in Section VII.A.13 for cl. 19, Brandstein's τ_i results from subtracting the distance between the source (*speech*) q and the i th sensor (*first physical microphone*), which is the *third distance*, from the distance between the source and a reference sensor (*second physical microphone*), which is the *fourth distance*, and then dividing by the speed of sound, c . Therefore, τ_i is *proportional to a quantity equal to a third distance subtracted from a fourth distance*. (*Id.*, ¶188.)

Claim 20 further recites an exponent *proportional to a sampling frequency multiplied by a quantity* such as Brandstein's τ_i because τ_i is *proportional to the sampling frequency multiplied by τ_i* , so that Brandstein's delay term $e^{-j\omega\tau_i}$ satisfies claim 20. (*Id.*)

Moreover, while Brandstein's delay term is expressed in seconds, the POSITA would have understood to express it in number of samples, as is common for digital signal processing involving discrete sampling. (*Id.*, ¶189.) To convert seconds to samples, the POSITA would have readily understood to multiply the delay (seconds) by the sampling frequency f_s (samples per second). (*Id.*, ¶190.) For example, Brandstein's delay term $e^{-j\omega\tau_i}$ becomes $e^{-j\omega\tau_i f_s}$. (*Id.*) Applying that exponential term delays a signal by $\tau_i f_s$ samples. (*Id.*)

A POSITA would have known to rewrite Brandstein’s near-field delay in terms of samples. (*Id.*, ¶191.) Brandstein itself motivates rewriting this delay element in terms of samples for use with the GJBF, as Fig. 5.1 is shown in terms of samples. (*Id.*; EX1005, 88-89 & n.1 (“ L_1 -sample delay”).) Brandstein also discloses implementing the GJBF with a DSP. (EX1005, 104, Fig. 5.16.) Additionally, digital signal processing was well-known, and a POSITA would have known to implement the GJBF digitally because digital circuits have advantages over analog systems with respect to manufacturing cost, flexibility of use, and accuracy. (EX1003, ¶191.) Converting Brandstein’s delay formula to samples would have involved simple and well-known operations. (*Id.*)

15. Independent Claim 1

Claim 1 is similar to independent claim 7 and dependent claim 8, with a few exceptions which are mapped below. *See* analysis for cls. 7-8 in Sections VII.A.3-4), which is incorporated herein. Specifically, the mappings of the following limitations apply to claim 1. (EX1003, ¶¶192-200.)

- [1pre] applies the mapping from [7pre] (Section VII.A.3.a);
- [1a] applies the mappings from [7a] and cl. 8 (Sections VII.A.3.b and VII.A.4);
- [1c] applies the mappings from [7a] and [7b] (Sections VII.A.3.b-c);

- [1d] applies the mappings from [7c], [7e]-[7f], and [7h] (Sections VII.A.3.d and VII.A.3.f-g); and
- [1f] applies the mappings from [7j]-[7n] (Sections VII.A.3.j-n).

(EX1003, ¶¶192-200.)

- a. **[1b] “a processing component coupled to the microphone array and generating a virtual microphone array including a first virtual microphone and a second virtual microphone,”**

Brandstein discloses *a virtual microphone array including a first virtual microphone and a second virtual microphone* for the same reasons discussed in Sections VII.A.3.b, VII.A.3.c regarding [7a] and [7b]. (EX1003, ¶¶194-195.)

It would have been obvious to digitally implement the GJBF using a processor (“*processing component*”), which was a well-known technique. (*Id.*, ¶195.) A POSITA would have understood that such a processor is *coupled* to the GJBF microphone array to process the microphone signals. (*Id.*)

- b. **[1e] “an adaptive noise removal application coupled to the processing component and generating denoised output signals by forming a plurality of combinations of signals output from the first virtual microphone and the second virtual microphone, by filtering and summing the plurality of combinations of signals in the time domain, and by a varying linear transfer**

function between the plurality of combinations of signals,”

Ikeda discloses this limitation for the same reasons discussed in Section VII.A.3.j-VII.A.3.m regarding [7j]-[7m]. (*Id.*, ¶¶198-199.)

A POSITA would have understood that Ikeda’s *adaptive noise removal application* is *coupled* to the *processing component* that generates the main and reference microphones. (EX1008, ¶¶ [0027]-[0029], Fig. 1; EX1003, ¶199)

16. Claim 2: “The system of claim 1, wherein the acoustic noise comprises noise content and the acoustic signals comprise speech content.”

Brandstein discloses this limitation for the same reasons discussed in Section VII.A.3.i regarding [7i]. (EX1003, ¶201.)

17. Claim 3: “The system of claim 2, wherein the speech content comprises human speech.”

Brandstein discloses this limitation for the same reasons discussed in Section VII.A.3.i regarding [7i]. (*Id.*, ¶202.)

18. Claim 5: “The system of claim 1 and further comprising: a communications channel coupled with the processing component and including one or more of a wireless channel, a wired channel, and a hybrid wireless/wired channel.”

Brandstein and Ikeda render this limitation obvious for the same reasons discussed in Section VII.A.6 regarding claim 11. (*Id.*, ¶203.)

19. Claim 6: “The system of claim 5 and further comprising: a communication device wirelessly coupled with the wireless channel of the communications channel.”

Brandstein and Ikeda render this limitation obvious for the same reasons discussed in Section VII.A.7 regarding claim 12. (*Id.*, ¶204.)

B. Ground 2: Brandstein in View of Ikeda and McCowan Render Obvious Claims 13, 19, and 20

This ground is the same as Ground 1, but additionally relies on McCowan’s near-field time-alignment calculations, as teaching the limitations in dependent claims 13, 19, and 20. (*Id.*, ¶¶205-213.)

1. Motivation to Combine

Applying near-field time-alignment for the GJBF’s bottom branch would have been part of a POSITA’s background knowledge because McCowan disclosed exactly that in 2002:

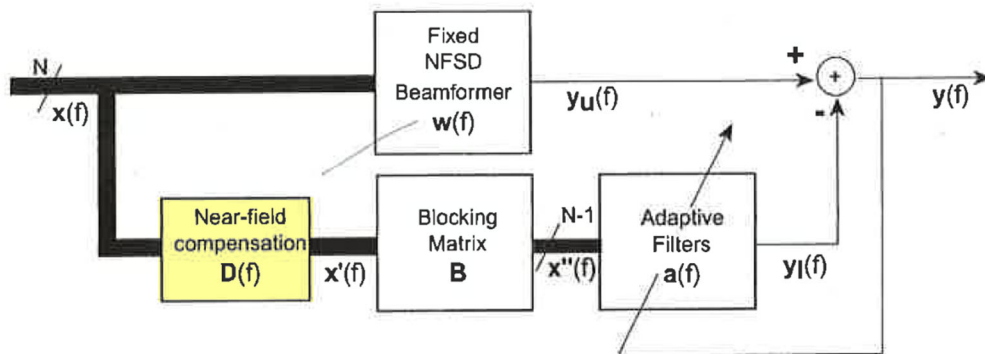


FIG. 3. Near-field adaptive beamformer.

(EX1009, 91, Fig. 3; EX1003, ¶206.)

McCowan discloses an adaptive noise cancellation system for near-field speech sources. (EX1009, 87-89, 97.) As shown in Fig. 3, the input $x(f)$ including signals from multiple microphones is processed via two signal paths. (*Id.*, 90, 93 (equation 15), 95-96.) The BM “block[s] the desired signal from the adaptive noise estimate” by subtracting one microphone signal from another. (*Id.*, 93, 95 (equations 18, 19); EX1003, ¶207.) As McCowan explains, “[t]o ensure complete blocking, the desired signal must both be time aligned and have equal amplitudes across all channels.” (EX1009, 93.)

For a near-field sound source, phase misalignment across microphones results from “[t]he difference in propagation time to [the i th] microphone with respect to the reference microphone ($i=1$),” defined as:

$$\tau_i = \frac{d_i - d_1}{c}, \quad (3)$$

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

$$\alpha_i = \frac{d_1}{d_i} \quad (4)$$

(*Id.*; EX1003, ¶208.)

McCowan explains that if $x_1(f)$ is defined as the desired source at a reference microphone, then the signal on the i th microphone is given by the equation:

$$x_i(f) = \alpha_i x_1(f) e^{-j2\pi f \tau_i} \quad (5)$$

(EX1009, 90.)

As shown in McCowan Fig. 3, “[o]nce this near-field compensation has been performed, a standard GJBF BM can be employed to block the desired signal from the adaptive path.” (*Id.*, 93; *see also id.*, 95 (“The blocking matrix and adaptive noise canceling filters are taken from the standard GSC technique [citation]” (citing Griffiths-Jim))); EX1003, ¶210.)

In view of this background knowledge, a POSITA would have known that some applications involve near-field target signals, that the modified GJBF’s BM requires time-alignment, that near-field time-alignment involves applying delays, and that McCowan discloses such near-field time-alignments followed by a basic GJBF. (EX1003, ¶211.) The motivation and knowledge to use near-field time-

alignment to implement the delays for the delay-and-subtract BM in the basic GJBF (as described in Brandstein’s Chapter 5) would have been immediately apparent to a POSITA, who “is also a person of ordinary creativity.” *KSR Int’l Co. v. Teleflex Inc.*, 550 U.S. 398, 421 (2007).

To the extent such near-field time-alignment would require further modifying Brandstein’s modified GJBF, the modification represents nothing more than the simple addition of one known element (near-field time alignment) to another known element (GJBF) to obtain the predictable results of effective target-signal blocking in the lower branch (to permit effective noise cancellation in the final output without canceling the target signal). (EX1003, ¶212.)

2. Reasonable Expectation of Success

The combination applies a known technique (time-alignment for near-field sources) to a well-known device (GJBF) that is ready for improvement, with the combination yielding predictable results. (*Id.*, ¶213.) A POSITA would have had a reasonable expectation of success because, as McCowan demonstrates, near-field time-alignment with the modified GJBF was trivial and known to work. (*Id.*)

3. **Claim 13: “The system of claim 7, wherein the second microphone signal is multiplied by a ratio, wherein the ratio is a ratio of a third distance to a fourth distance, the third distance being between the first physical microphone and the speech source and the fourth distance being between the second physical microphone and the speech source.”**

Brandstein in combination with McCowan renders this limitation obvious.

(EX1003, ¶¶214-215.)

As discussed above, the microphone attenuation factor α_i in McCowan equation (4):

$$\alpha_i = \frac{d_1}{d_i}, \quad (4)$$

is a ratio of d_1 (the distance between the speech source and a reference (*first*) microphone) to d_i (the distance between the speech source and the i th (*e.g., second*) microphone). (*Id.*, ¶215.) This attenuation factor is *multiplied* to the i th (*e.g., second*) microphone signal to achieve equal amplitudes across all channels. (*Id.*)

4. **Claim 19: “The system of claim 18, wherein the delay is raised to a power that is proportional to a time difference between arrival of the speech at the first virtual microphone and arrival of the speech at the second virtual microphone.”**

Brandstein in combination with McCowan discloses this limitation. (*Id.*,

¶¶216-217.)

McCowan defines τ_i as:

$$\tau_i = \frac{d_i - d_1}{c}, \quad (3)$$

(EX1009, 89.) In the above equations, c is the speed of sound. For McCowan, d_1 is the distance from the *speech source* to the reference (*first*) microphone and d_i is the distance from the speech source to the i th (*second*) microphone. (EX1003, ¶217.) The difference between d_i and d_1 is divided by the speed of sound to obtain *a time difference between arrival of the speech at the first virtual microphone and arrival of the speech at the second virtual microphone, τ_i* . (*Id.*)

5. **Claim 20: “The system of claim 19, wherein the power is proportional to a sampling frequency multiplied by a quantity equal to a third distance subtracted from a fourth distance, the third distance being between a first physical microphone and the speech source, the fourth distance being between a second physical microphone and the speech source, and the first and second physical microphones are positioned in the microphone array.”**

Brandstein in combination with McCowan discloses this limitation. (*Id.*, ¶¶218-219.)

Claim 20 further recites an exponent *proportional to a sampling frequency multiplied by a quantity* such as McCowan’s τ_i . (*Id.*) McCowan’s τ_i is proportional to the sampling frequency multiplied by τ_i , so McCowan’s delay term $e^{-j\omega\tau_i}$

satisfies claim 20 for the same reasons as discussed above in Section VII.A.14

(Ground 1, cl. 20). (*Id.*)

C. Ground 3: Brandstein in View of Ikeda and Yang Render Obvious Claims 4, 10, and 15

This ground is the same as Ground 1, but additionally relies on Yang’s VAD, as teaching the limitations in dependent claims 4, 10, and 15. (EX1003, ¶¶220-223.)

1. Motivation to Combine

It would have been obvious to incorporate a VAD into the Brandstein-Ikeda combination. (EX1003, ¶221.) Specifically, the POSITA would have been motivated to apply the teachings of Yang’s VAD, which determines the “timing at which the microphone units M1 and M2 do not pick up the target sound,” such as human speech, and provides a control signal that enables the adaptation of the coefficients of the [adaptive filter 450] when no speech activity is detected.” (EX1008, ¶¶[0028], [0036]; EX1010, ¶¶[0058]-[0062], Fig. 4; EX1003, ¶221).

As Yang teaches and the POSITA would have appreciated, coupling a VAD to Ikeda’s adaptive filter would have enhanced the filtering. (EX1003, ¶221.) Specifically, Ikeda’s adaptive filter 7 has response characteristics that depend on the timing when the microphones “do not pick up the target sound.” (EX1008, ¶[0028].) Because a VAD provides signals indicating the timing of when speech is picked up or not, incorporating a VAD would have improved Ikeda’s adaptive filtering by providing accurate timing signals. (EX1003, ¶221.)

A POSITA would have recognized the benefit of using a VAD to identify the presence or absence of speech to improve adaptation for noise cancellation. (*Id.*) This modification would have been nothing more than combining prior art elements according to known methods to yield predictable results. (*Id.*) Indeed, Yang states that “detection of speech activity can be performed in various manners known in the art.” (EX1010, [¶0031].)

2. Reasonable Expectation of Success

A POSITA would have had a reasonable expectation of success to add Yang’s VAD to the Brandstein-Ikeda combination. (EX1003, ¶¶222-223.) Yang teaches that the VAD can be implemented with a GJBF. (EX1010, ¶¶[0026], [0038]; EX1003, ¶222.) Furthermore, a POSITA would have recognized that the adaptive noise reduction system of Yang’s Fig. 4 takes as inputs a speech plus noise signal and a mostly noise only signal, which are the same types of signals output by the top and bottom GJBF branches, respectively. (EX1010, ¶¶[0058]-[0062], Fig. 4; EX1003, ¶223.) Therefore, Yang itself suggests combining a VAD with a GJBF, and using a VAD to determine the timing when microphones do not pick up human speech would have been within the knowledge and skill of a POSITA. (EX1003, ¶223.)

3. Claim 10: “The system of claim 7 and further comprising: a voice activity detector (VAD) coupled with the processing component and operative to generate voice activity signals.”

Brandstein, Ikeda and Yang render obvious this limitation. (*Id.*, ¶¶224-228.)

As explained for claim 11 in Ground 1, Ikeda discloses a *processing component* as a DSP that performs adaptive filtering. (EX1008, ¶¶[0016]-[0017], [0030]; EX1003, ¶¶225-226.)¹¹ It would have been obvious to incorporate Yang’s VAD to be *coupled with* Ikeda’s *processing component* for improving the functionality of the adaptive filtering. (EX1008, ¶¶[0016]-[0017], [0028], [0030], [0036]; EX1010, ¶¶[0058]-[0062], Fig. 4; EX1003, ¶227). Yang’s VAD detects voice activity and is therefore *operative to generate voice activity signals*. (EX1003, ¶228.)

4. Claim 15: “The system of claim 14 and further comprising: a voice activity detector (VAD) coupled with the processing component and operative to generate voice activity signals.”

Brandstein, Ikeda, and Yang render obvious this limitation for the same reasons discussed in Section VII.C.3 regarding claim 10. (*Id.*, ¶229.)

¹¹ “[T]he processing component” element of claim 10 lacks antecedent basis. For purposes of this Petition, Petitioner assumes that claim 10 requires “a processing component.”

5. **Claim 4: “The system of claim 1 and further comprising: a voice activity detector (VAD) coupled with the processing component and operative to generate voice activity signals.”**

Brandstein, Ikeda, and Yang render obvious this limitation for the same reasons discussed in Section VII.C.3 regarding claim 10. (EX1003, ¶230.)

VIII. SECONDARY CONSIDERATIONS OF NONOBVIOUSNESS

Secondary considerations should be considered but do not control the obviousness conclusion. *Newell Cos. v. Kenney Mfg. Co.*, 864 F.2d 757, 768 (Fed. Cir. 1988). Where, as here, a strong *prima facie* obviousness showing exists, even relevant secondary considerations supported by substantial evidence may not dislodge the obviousness conclusion. *Leapfrog Enters. v. Fisher-Price, Inc.*, 485 F.3d 1157, 1162 (Fed. Cir. 2007). Petitioner is aware of no evidence supporting a claim for secondary considerations.

IX. INSTITUTION IS APPROPRIATE UNDER §314(A).

A. Co-Pending Litigation Does Not Warrant Discretionary Denial (*Fintiv*)

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

Factor 1 (stay) favors institution. Two other Jawbone cases involving the same patents were previously transferred from Texas to Northern California (“NDCA”), and Meta has moved to transfer venue to NDCA on similar facts. *In re*

Meta Platforms, Inc. v. Jawbone Innovations, LLC, IPR2024-00350

IPR Petition – U.S. Patent No. 10,779,080

Google LLC, 58 F.4th 1379, 1383 (Fed. Cir. 2023); *Jawbone Innovations, LLC v. Amazon.com, Inc., et al.*, No. 2:21-cv-00435-JRG, Dkt. No. 60 (E.D. Tex. Sept. 21, 2022).

NDCA routinely grants motions to stay when IPRs have been instituted. *E.g.*, *Uniloc USA Inc. v. LG Elecs., U.S.A. Inc.*, No. 18-cv-06737-JST, 2019 WL 1905161 (N.D. Cal. April 29, 2019); *Symantec Corp. v. Zscaler, Inc.*, No. 17-cv-04426-JST, 2018 WL 3539267 (N.D. Cal. July 23, 2018); *Twilio, Inc. v. TeleSign Corp.*, No. 16-cv-06925-LHK, 2018 WL 1609630 (N.D. Cal. April 3, 2018); *Robert Bosch Healthcare Sys., Inc. v. Cardiocom, LLC*, No. C-14-1575 EMC, 2014 WL 3107447 (N.D. Cal. July 3, 2014). Indeed, both prior NDCA Jawbone cases were stayed pending outcome of IPRs. *Jawbone Innovations, LLC v. Amazon.com, Inc. et al.*, No. 3:22-cv-06727-TLT, Dkt. 105 (N.D. Cal. Mar. 22, 2023); *Jawbone Innovations, LLC v. Google LLC*, No. 3:23-cv-00466-TLT, Dkt. 137 (N.D. Cal. Apr. 27, 2023).

While those IPRs are now terminated pursuant to settlements, institution decisions are imminent on new IPR petitions challenging each patent-at-issue in the litigation between Jawbone and Petitioner. *See infra*, Section XI.B. Further, even if the case is not transferred, Judge Albright regularly enters stays when all patents are subject to instituted IPRs. *See Lone Star SCM Sys., Ltd. v. Zebra Tech. Corp.*, No. 6:21-cv-00842-ADA, 2023 WL 4923962 (W.D. Tex. Jul. 31, 2023); *Sonrai Memory Ltd. v. Western Digital Tech., Inc.*, No. 6:21-cv-01168-ADA, 2922 WL 3108818 (W.D.

Tex. Aug. 4, 2022); *Xylon Licensing LLC v. Lone Star Nat’l Bancshares-Texas, Inc.*, No. 6:21-cv-00302-ADA, 2022 WL 2078030 (W.D. Tex. Jun. 8, 2022).

Because nearly all of the petitions rely upon the same grounds as a previously instituted proceeding that has since been terminated based on settlement, institution and a litigation stay is highly likely.

Factor 2 (proximity to trial) favors institution. Should the Court adopt the parties’ agreed schedule in the co-pending litigation, trial will be set for June 11, 2025, in or around the same time as the projected statutory deadline. *Fintiv*, IPR2020-00019, Paper 11 at 9 (“If the court’s trial date is at or around the same time as the projected statutory deadline or even significantly after the projected statutory deadline, the decision whether to institute will likely implicate other factors”).

Factor 3 (investment in parallel litigation) favors institution. The co-pending litigation is still in its early stages. The Court has yet to adopt a case schedule. The parties have not conducted any fact discovery, exchanged proposed claim constructions, submitted claim construction briefs, nor served expert reports. Indeed, Petitioner is filing this petition soon after the identification of asserted claims and well before serving preliminary invalidity contentions currently due on February 25, 2024. *Dish Network, LLC v. Broadband iTV, Inc.*, IPR2020-01267, Paper 15, at 20-21 (P.T.A.B. Jan. 21, 2021) (finding diligence when petition filed less than three months from notice of asserted claims); *Apple Inc. v. Parus Holdings, Inc.*, IPR2020-

00686, Paper 9 at 13 (P.T.A.B. Sept. 23, 2020) (finding “Petitioner acted expeditiously” in filing “shortly after serving its preliminary invalidity contentions” and “prior to the submission of claim construction positions”). Importantly, the “remaining investment” outweighs any investments made. *Samsung Elecs. Am. Inc. v. Snik LLC*, IPR2020-01428, Paper 10, 11 (P.T.A.B. Mar. 9, 2021); *Huawei Techs. Co. v. WSOU Invs., LLC*, IPR2021-00226, Paper 10, 13-14 (P.T.A.B. Jun. 10, 2021).

Factor 4 (overlapping issues) favors institution. If this IPR is instituted and the case is not transferred, Petitioner will not pursue the petition’s grounds before the district court by stipulation. *Sotera Wireless, Inc. v. Masimo Corp.*, IPR2020-01019, Paper 12 (P.T.A.B. Dec. 1, 2020); *Sand Revolution II, LLC v. Cont’l Intermodal Grp.-Trucking LLC*, IPR2019-01393, Paper 24, 11-12 (P.T.A.B. Jun. 16, 2020); *Google LLC v. Jawbone Innovations, LLC*, IPR2022-00630, Paper 10, 16-17 (P.T.A.B. Sept. 13, 2022).

Factor 5 (same parties) is neutral. This factor weighs “either in favor of, or against, exercising discretion to deny institution depending on which tribunal was likely to address the challenged patent first.” *Parus*, IPR2020-00686, Paper 9 at 19; *see Google*, IPR2022-00630, Paper 10, 17. Here, the close proximity of the tentative trial date neither weighs in favor or against.

Factor 6 (other considerations) favors institution. Because the challenged claims fail to limit the generation of virtual microphones to a specific formula, they encompass prior art developed in the 1980s, and any differences are obvious. This petition’s merits are therefore particularly strong.

Accordingly, *Fintiv* favors institution.

B. Existence of Previously Filed IPRs Does Not Warrant Discretionary Denial (*General Plastic*)

IPRs against the same claims of the ’080 Patent have been filed by Google (IPR2022-01059, May 27, 2022), Amazon (IPR2023-00283, Nov. 28, 2022), Samsung (IPR2022-01320, July 27, 2022), and Sony (IPR2023-01065, June 16, 2023). These petitions do not warrant discretionary denial under *General Plastic Indus. Co., Ltd. v. Canon Kabushiki Kaisha*, IPR2016-01357, Paper 19, 15-16 (P.T.A.B. Sept. 6, 2017) (precedential).

Meta and the prior petitioners are different parties and have no significant relationship. They are not co-defendants. They are competitors accused of infringement—in different actions pending in different courts—based on sales of different products. This weighs against denial. *NetNut Ltd. v. Bright Data Ltd.*, IPR2021-00465, Paper 11, 9 (P.T.A.B. Aug. 12, 2021); *Mercedes-Benz USA, LLC v. Carucel Invs. L.P.*, IPR2019-01404, Paper 12, 11-12 (P.T.A.B. Jan. 22, 2020);

Meta Platforms, Inc. v. Jawbone Innovations, LLC, IPR2024-00350
IPR Petition – U.S. Patent No. 10,779,080

Toshiba Am. Info. Sys., Inc. v. Wallelex Microelecs. Ltd., IPR2018-01538, Paper 11, 20 (P.T.A.B. Mar. 5, 2019).

The remaining factors consider timing issues, *e.g.*, when Petitioner became aware of prior art, whether the PO or Board responded to prior petitions, and the length of time that has passed. But the sequential IPR filings here were caused by PO's sequential district court filings and should not be held against Petitioner. Specifically, PO sued Samsung in May 2021 (2:21-cv-00186-JRG-RSP (E.D. Tex.)), waited four months before suing Google and Apple (6:21-cv-00985-ADA, 6:21-cv-00984-ADA (W.D. Tex.)), another two months to sue Amazon, and then waited another fourteen months, until Feb. 28, 2023, to sue Meta, Sony, and LG. The Board should not deny this petition merely because PO sued other defendants earlier and independently. *Toshiba*, IPR2018-01538, Paper 11, 21-22 (PO's complaint about multiple IPRs "is not persuasive when the volume appears to be a direct result of its own litigation activity"); *id.*, 22.

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

X. INSTITUTION IS APPROPRIATE UNDER §325(D).

This petition should not be discretionarily denied under § 325(d). *Advanced Bionics, LLC v. MED-EL Elektromedizinische Geräte GmbH*, IPR2019-01469,

Paper 6 (Feb. 13, 2020) (precedential). None of the prior art asserted in this petition was considered by the examiner during prosecution, and this petition’s arguments are not substantially the same as or cumulative of the arguments previously presented to the Office during original prosecution.

XI. MANDATORY NOTICES

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

Meta Platforms, Inc. is the real party-in-interest.

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

To the best knowledge of Petitioner, the ’080 Patent is or has been involved in the following district court litigations and petitions for *inter partes* review:

Name	Number	Court	Filed
<i>Jawbone Innovations, LLC v. Amazon.com, Inc.</i>	2:22-cv-06727	N.D. Cal.	Nov. 29, 2021 (transferred from E.D. Tex. on Nov. 1, 2022, No. 2:21-cv-00435))
<i>Jawbone Innovations, LLC v. Apple Inc.</i>	6:21-cv-00984	W.D. Tex.	Sept. 23, 2021
<i>Jawbone Innovations, LLC v. Google LLC</i>	3:23-cv-00466	N.D. Cal.	Sept. 23, 2021 (transferred from W.D. Tex. on Feb. 1, 2023, No. 6:21-cv-00985)
<i>Jawbone Innovations, LLC v. Samsung Electronics Co., Ltd.</i>	2:21-cv-00186	E.D. Tex.	May 27, 2021
Petition for <i>Inter Partes</i>	IPR2022-01320	P.T.A.B.	Jul. 27, 2022

Name	Number	Court	Filed
Review [Samsung & Apple]			
Petition for <i>Inter Partes</i> Review [Google]	IPR2022-01059	P.T.A.B.	May 27, 2022
Petition for <i>Inter Partes</i> Review [Amazon]	IPR2023-00284	P.T.A.B.	Nov. 28, 2022
<i>Jawbone Innovations, LLC v. Meta Platforms, Inc. d/b/a/ Meta</i>	6:23-cv-00158	W.D. Tex.	Feb. 28, 2023
<i>Jawbone Innovations, LLC v. ZTE Corp.</i>	2:23-cv-00082	E.D. Tex.	Feb. 28, 2023
<i>Jawbone Innovations, LLC v. Panasonic Holdings Corp.</i>	2:23-cv-00081	E.D. Tex.	Feb. 28, 2023
<i>Jawbone Innovations, LLC v. Guangdong Oppo Mobile Tele. Corp., Ltd. d/b/a OPPO</i>	2:23-cv-00079	E.D. Tex.	Feb. 28, 2023
<i>Jawbone Innovations, LLC v. LG Electronics, Inc.</i>	2:23-cv-00078	E.D. Tex.	Feb. 28, 2023
<i>Jawbone Innovations, LLC v. HTC Corp.</i>	2:23-cv-00077	E.D. Tex.	Feb. 28, 2023
<i>Jawbone Innovations, LLC v. Sony Elec., Inc.</i>	2:23-cv-01161	D.N.J.	Feb. 28, 2023
Petition for <i>Inter Partes</i> Review [Sony]	IPR2023-01065	P.T.A.B.	Jun. 15, 2023

In addition, institution decisions are imminent on new IPR petitions challenging each of the patents at issue in the district court litigation between Jawbone and Petitioner.

- IPR2023-01248, Paper 3 (P.T.A.B. July 26, 2023) (’543 Patent, claims 1-26);
- IPR2023-01222, Paper 1 (P.T.A.B. July 20, 2023) (’543 Patent, claims 1-26);

- IPR2023-01128, Paper 1 (P.T.A.B. June 27, 2023) (U.S. Pat. No. 8,326,611, claims 1-2, 5-28);
- IPR2023-01129, Paper 1 (P.T.A.B. June 27, 2023) (U.S. Pat. No. 8,326,611, claims 29, 31-44);
- IPR2023-01176, Paper 1 (P.T.A.B. July 6, 2023) (U.S. Pat. No. 8,326,611, claims 29-44);
- IPR2023-01177, Paper 1 (P.T.A.B. July 6, 2023) (U.S. Pat. No. 8,326,611, claims 1-28);
- IPR2023-01155, Paper 2 (P.T.A.B. July 6, 2023) (U.S. Pat. No. 8,326,611, claims 1-28);
- IPR2023-01156, Paper 1 (P.T.A.B. July 6, 2023) (U.S. Pat. No. 8,326,611, claims 29-44);
- IPR2023-01065, Paper 1 (P.T.A.B. June 15, 2023) (U.S. Pat. No. 10,779,080, claims 1-20);
- IPR2023-01110, Paper 2 (P.T.A.B. June 23, 2023) (U.S. Pat. No. 7,246,058, claims 1-5);
- IPR2023-01119, Paper 1 (P.T.A.B. June 23, 2023) (U.S. Pat. No. 7,246,058, claims 1-5);
- IPR2023-01130, Paper 1 (P.T.A.B. June 27, 2023) (U.S. Pat. No. 11,122,357, claims 1-20);
- IPR2023-01134, Paper 2 (P.T.A.B. June 29, 2023) (U.S. Pat. No. 11,122,357, claims 1-20);
- IPR2023-01153, Paper 1 (P.T.A.B. June 27, 2023) (U.S. Pat. No. 11,122,357, claims 1-20);
- IPR2023-01127, Paper 1 (P.T.A.B. June 27, 2023) (U.S. Pat. No. 8,019,091, claims 1-20);
- IPR2023-01154, Paper 2 (P.T.A.B. July 5, 2023) (U.S. Pat. No. 8,019,091, claims 1-20);

- IPR2023-01166, Paper 1 (P.T.A.B. July 5, 2023) (U.S. Pat. No. 8,019,091, claims 1-20);
- IPR2023-01083, Paper 1 (P.T.A.B. June 23, 2023) (U.S. Pat. No. 8,321,213, claims 1, 3-13);
- IPR2023-01084, Paper 1 (P.T.A.B. June 23, 2023) (U.S. Pat. No. 8,321,213, claims 14-36, 39-42);
- IPR2023-01111, Paper 2 (P.T.A.B. June 23, 2023) (U.S. Pat. No. 8,321,213, claims 1-13);
- IPR2023-01112, Paper 2 (P.T.A.B. June 23, 2023) (U.S. Pat. No. 8,321,213, claims 14-42);
- IPR2023-01117, Paper 1 (P.T.A.B. June 23, 2023) (U.S. Pat. No. 8,321,213, claims 14-42);
- IPR2023-01118, Paper 1 (P.T.A.B. June 23, 2023) (U.S. Pat. No. 8,321,213, claims 1-13);
- IPR2023-01228, Paper 1 (P.T.A.B. July 20, 2023) (U.S. Pat. No. 8,503,691, claims 1-22, 29-40);
- IPR2023-01229, Paper 1 (P.T.A.B. July 24, 2023) (U.S. Pat. No. 8,503,691, claims 23-28, 41-46);
- IPR2023-01291, Paper 1 P.T.A.B. August 31, 2023) (U.S. Pat. No. 8,467,543)
- IPR2023-01352, Paper 1 (P.T.A.B. August 31, 2023) (U.S. Pat. No. 7,246,058).

C. Lead and Back-Up Counsel Under 37 C.F.R. § 42.8(b)(3)

Meta provides the following designation of counsel:

Lead Counsel	Backup Counsel
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	Telephone: (857) 353-4511
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Pursuant to 37 C.F.R. § 42.10(b), a Power of Attorney accompanies this petition.

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

Service information for counsel is provided above. Meta consents to service by email at: AO_Meta_Jawbone@AllenOvery.com.

XII. GROUNDS FOR STANDING

Meta certifies that the '080 Patent is available for IPR; Meta is not barred or estopped from requesting IPR; and this petition is being filed within one year of service of the complaint in the co-pending district court litigation.

XIII. CONCLUSION

Petitioner requests institution of this IPR and cancellation of the claims.

Respectfully submitted,

December 20, 2023

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CLAIM LISTING APPENDIX

[1Pre.] A system, comprising:

[1a.] a microphone array including a first physical microphone outputting a first microphone signal and a second physical microphone outputting a second microphone signal;

[1b.] a processing component coupled to the microphone array and generating a virtual microphone array including a first virtual microphone and a second virtual microphone,

[1c.] the first virtual microphone including a first combination of the first microphone signal and the second microphone signal, the second virtual microphone including a second combination of the first microphone signal and the second microphone signal, wherein the second combination is different from the first combination,

[1d.] wherein the first virtual microphone and the second virtual microphone have substantially similar responses to noise and substantially dissimilar responses to speech; and

[1e.] an adaptive noise removal application coupled to the processing component and generating denoised output signals by forming a plurality of combinations of signals output from the first virtual microphone and the second virtual microphone, by filtering and summing the plurality of combinations of signals in the time domain, and by a varying linear transfer function between the plurality of combinations of signals,

[1f.] wherein the denoised output signals include less acoustic noise than acoustic signals received at the microphone array.

[2.] The system of claim 1, wherein the acoustic noise comprises noise content and the acoustic signals comprise speech content.

[3.] The system of claim 2, wherein the speech content comprises human speech.

[4.] The system of claim 1 and further comprising: a voice activity detector (VAD) coupled with the processing component and operative to generate voice activity signals.

[5.] The system of claim 1 and further comprising: a communications channel coupled with the processing component and including one or more of a wireless channel, a wired channel, and a hybrid wireless/wired channel.

[6.] The system of claim 5 and further comprising: a communication device wirelessly coupled with the wireless channel of the communications channel.

[7pre.] A system, comprising:

[7a.] a first virtual microphone formed from a first combination of a first microphone signal and a second microphone signal, wherein the first microphone signal is generated by a first physical microphone and the second microphone signal is generated by a second physical microphone;

[7b.] a second virtual microphone formed from a second combination of the first microphone signal and the second microphone signal, wherein the second combination is different from the first combination,

[7c.] wherein the first virtual microphone has a first linear response to speech and first linear response to noise,

[7d.] the first linear response to speech being substantially similar across a plurality of frequencies for a speech source located within a predetermined angle relative to an axis of the microphone array and devoid of a null,

[7e.] wherein the second virtual microphone has a second linear response to speech that has a single null oriented in a direction toward a source of the speech and a second linear response to noise,

[7f.] wherein the second linear response to noise is substantially similar to the first linear response to noise,

[7g.] one or both of the first linear response to noise and the second linear response to noise being non-zero in a direction toward a source of noise, and

[7h.] the second linear response to speech is substantially dissimilar to the first linear response to speech,

[7i.] wherein the speech is human speech; and

[7j.] an adaptive noise removal application coupled to the first and second virtual microphones and generating denoised output signals

[7k.] by forming a plurality of combinations of signals output from the first virtual microphone and the second virtual microphone,

[7l.] by filtering and summing the plurality of combinations of signals in the time domain, and

[7m.] by a varying linear transfer function between the plurality of combinations of signals,

[7n.] wherein the denoised output signals include less acoustic noise than acoustic signals received at the first and second physical microphones.

[8.] The system of claim 7 and further comprising:
a microphone array, the first and second physical microphones positioned in the microphone array.

[9.] The system of claim 7, wherein the single null is a region of the second linear response to speech having a measured response level that is lower than the measured response level of any other region of the second linear response to speech.

[10.] The system of claim 7 and further comprising: a voice activity detector (VAD) coupled with the processing component and operative to generate voice activity signals.

[11.] The system of claim 7 and further comprising: a communications channel coupled with the processing component and including one or more of a wireless channel, a wired channel, and a hybrid wireless/wired channel.

[12.] The system of claim 11 and further comprising: a communication device wirelessly coupled with the wireless channel of the communications channel.

[13.] The system of claim 7, wherein the second microphone signal is multiplied by a ratio, wherein the ratio is a ratio of a third distance to a fourth distance, the third distance being between the first physical microphone and the speech source and the fourth distance being between the second physical microphone and the speech source.

[14Pre.] A system, comprising:

[14a.] a first virtual microphone comprising a first combination of a first microphone signal and a second microphone signal,

[14b.] the first virtual microphone having a first linear response to speech and a first linear response to noise,

[14c.] the first linear response to speech being substantially similar across a plurality of frequencies for a speech source located within a predetermined angle relative to an axis of a microphone array,

[14d.] wherein the first microphone signal is output from a first physical microphone and the second microphone signal is output from a second physical microphone;

[14e.] a second virtual microphone comprising a second combination of the first microphone signal and the second microphone signal,

[14f.] the second virtual microphone having a second linear response to speech and a second linear response to noise,

[14g.] the second linear response to noise being substantially similar to the first linear response to noise,

[14h.] one or both of the first linear response to noise and the second linear response to noise being non-zero in a direction toward a source of noise, and

[14i.] the second linear response to speech being substantially dissimilar to the first linear response to speech,

[14j.] wherein the second combination is different from the first combination,

[14k.] wherein the first virtual microphone and the second virtual microphone are distinct virtual directional microphones; and

[14l.] a processing component coupled to the first and second virtual microphones, the processing component including an adaptive noise removal application receiving acoustic signals from the first virtual microphone and the second virtual microphone, filtering and summing the acoustic signals in the time domain, applying a varying linear transfer function between the acoustic signals, and generating an output signal,

[14m.] wherein the output signal is a denoised acoustic signal.

[15.] The system of claim 14 and further comprising: a voice activity detector (VAD) coupled with the processing component and operative to generate voice activity signals.

[16.] The system of claim 14 and further comprising: a communications channel coupled with the processing component and including one or more of a wireless channel, a wired channel, and a hybrid wireless/wired channel.

[17.] The system of claim 16 and further comprising: a communication device wirelessly coupled with the wireless channel of the communications channel.

[18.] The system of claim 14, wherein the acoustic signals from the first virtual microphone, the second virtual microphone or both are delayed.

[19.] The system of claim 18, wherein the delay is raised to a power that is proportional to a time difference between arrival of the speech at the first virtual microphone and arrival of the speech at the second virtual microphone.

[20.] The system of claim 19, wherein the power is proportional to a sampling frequency multiplied by a quantity equal to a third distance subtracted from a fourth distance, the third distance being between a first physical microphone and the speech source, the fourth distance being between a second physical microphone and the speech source, and the first and second physical microphones are positioned in the microphone array.

CERTIFICATE OF COMPLIANCE

Pursuant to 37 C.F.R. § 42.24(d), the undersigned certifies that foregoing
**PETITION FOR INTER PARTES REVIEW OF U.S. PATENT NO.
10,779,080**, exclusive of the parts exempted as provided in 37 C.F.R. §42.24(a),
contains 13,858 words and therefore complies with the type-volume limitations of
37 C.F.R. §42.24(a).

Respectfully submitted,

December 20, 2023

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Counsel for Petitioner,
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CERTIFICATE OF SERVICE

I hereby certify that a true and correct copy of the foregoing **PETITION FOR INTER PARTES REVIEW OF U.S. PATENT NO. 10,779,080 AND EXHIBITS 1001-1023** are being served on December 20, 2023, via Federal Express overnight mail on counsel of record for U.S. Patent No. 10,779,080 as addressed below:

Mark Leonardo
Nutter McClennen & Fish LLP
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155 Seaport Boulevard
Boston, MA 02210-2604

A courtesy copy is also being served on counsel for the patent holder in the pending litigation *Jawbone Innovations, LLC v. Meta Platforms, Inc.*, 6:23-cv-00158 (W.D. Tex., filed February 28, 2023):

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IPR Petition – U.S. Patent No. 10,779,080

Respectfully submitted,

December 20, 2023

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