

Filed: June 27, 2023

Filed on behalf of Meta Platforms, Inc.

By: Lisa Nguyen  
David Tennant  
Eric E. Lancaster  
Daniel Margolis  
Grace Wang  
Alan Billharz  
Colby Davis  
Allen & Overy LLP  
550 High Street  
Palo Alto, CA 94301  
Telephone: 650-388-1724  
Facsimile: 202-683-3999  
Email: AO\_Meta\_Jawbone@AllenOvery.com

UNITED STATES PATENT AND TRADEMARK OFFICE  
BEFORE THE PATENT TRIAL AND APPEAL BOARD

---

**META PLATFORMS, INC.,**  
Petitioner,

v.

**JAWBONE INNOVATIONS, LLC,**  
Patent Owner.

---

IPR2023-01130  
U.S. Patent No. 11,122,357

---

**PETITION FOR *INTER PARTES* REVIEW OF CLAIMS 1-20 OF  
U.S. PATENT NO. 11,122,357 PURSUANT TO  
35 U.S.C. §§ 311–319, 37 C.F.R. § 42**

**TABLE OF CONTENTS**

|                                                                                                                                                                                     |    |
|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|----|
| INTRODUCTION-----                                                                                                                                                                   | 1  |
| I. BACKGROUND -----                                                                                                                                                                 | 3  |
| A. Griffiths and Jim Publish Their Seminal GSC Article in<br>1982. -----                                                                                                            | 3  |
| B. Over the Next Two Decades, the GSC was Used in<br>Microphone Arrays to Reduce Noise. -----                                                                                       | 6  |
| C. Published in 2001, Brandstein Illustrates How to Use a GSC<br>with a Microphone Array to Reduce Noise. -----                                                                     | 7  |
| D. Contemporaneously, Gannot Taught Adapting the GSC to<br>Handle Arbitrary Transfer Functions. -----                                                                               | 10 |
| E. Other Concepts in the '357 Patent Were Well Known. -----                                                                                                                         | 13 |
| 1. Filtering and Summing in the Time Domain Were<br>Well Known. -----                                                                                                               | 13 |
| 2. Delaying Signals Based on Geometry to Adjust for<br>Differences in Arrival Times Was Well Known. -----                                                                           | 14 |
| II. THE '357 PATENT -----                                                                                                                                                           | 16 |
| A. The '357 Patent Discloses Nothing Innovative.-----                                                                                                                               | 16 |
| 1. The '357 Patent Purports to Distinguish Itself from<br>the Prior Art by Using a Virtual Microphone<br>Designed to Capture Only Noise, Which Had Been<br>Known for Decades. ----- | 16 |
| 2. The '357 Patent Concedes It Relies on Known<br>Techniques to Form Virtual Microphones from<br>Physical Microphones.-----                                                         | 16 |

|      |                                                                                                                                                         |    |
|------|---------------------------------------------------------------------------------------------------------------------------------------------------------|----|
| 3.   | The '357 Patent Discloses Formulas for the<br>Purportedly Innovative Set of Virtual Microphones,<br>But These Formulas Rely on Near-Field Design. ----- | 18 |
| B.   | The Claims Recite Generic Virtual Microphones and<br>Generic Signal Processing. -----                                                                   | 20 |
| C.   | The Claims Were Not Carefully Scrutinized During<br>Prosecution. -----                                                                                  | 21 |
| III. | STATEMENT OF RELIEF REQUESTED -----                                                                                                                     | 22 |
| A.   | Grounds -----                                                                                                                                           | 22 |
| B.   | The Earliest Priority Date the '357 Patent Claims Is June<br>13, 2007.-----                                                                             | 22 |
| C.   | The References Are Prior Art.-----                                                                                                                      | 23 |
| D.   | The Asserted References Are Analogous Art. -----                                                                                                        | 23 |
| IV.  | LEVEL OF ORDINARY SKILL-----                                                                                                                            | 24 |
| V.   | CLAIM CONSTRUCTION -----                                                                                                                                | 24 |
| VI.  | GROUND OF UNPATENTABILITY -----                                                                                                                         | 25 |
| A.   | Ground 1: Brandstein and Gannot -----                                                                                                                   | 25 |
| 1.   | Claim 1 -----                                                                                                                                           | 25 |
| a.   | Preamble -----                                                                                                                                          | 25 |
| b.   | First Virtual Microphone Comprising a<br>Combination of Signals from First and Second<br>Physical Microphones-----                                      | 26 |
| c.   | Second Virtual Microphone -----                                                                                                                         | 27 |
| d.   | Substantially Similar Responses to Noise and<br>Substantially Dissimilar Responses to Speech-----                                                       | 28 |

|     |                                                                                                                    |    |
|-----|--------------------------------------------------------------------------------------------------------------------|----|
| e.  | A Signal Processor Operative to Combine<br>Microphone Signals by Filtering and Summing<br>in the Time Domain ----- | 31 |
| f.  | Applying a Varying Linear Transfer Function-----                                                                   | 33 |
| g.  | Generating an Output Signal with Reduced<br>Noise -----                                                            | 36 |
| 2.  | Claim 2 -----                                                                                                      | 37 |
| 3.  | Claim 3 -----                                                                                                      | 38 |
| 4.  | Claim 4 -----                                                                                                      | 39 |
| 5.  | Claim 5 -----                                                                                                      | 41 |
| a.  | Claim 5 Encompasses Standard Near-Field<br>Time-Alignment. -----                                                   | 42 |
| b.  | Brandstein Discloses or Renders Obvious<br>Standard Near-Field Time-Alignments for the<br>GSC. -----               | 43 |
| 6.  | Claim 6 -----                                                                                                      | 49 |
| 7.  | Claim 7 -----                                                                                                      | 52 |
| 8.  | Claim 8 -----                                                                                                      | 52 |
| 9.  | Claim 9 -----                                                                                                      | 53 |
| 10. | Claim 10 -----                                                                                                     | 53 |
| 11. | Claim 11 -----                                                                                                     | 55 |
| 12. | Claim 12 -----                                                                                                     | 56 |
| 13. | Claim 13 -----                                                                                                     | 57 |
| 14. | Claim 14 -----                                                                                                     | 59 |

|       |                                                                                                   |    |
|-------|---------------------------------------------------------------------------------------------------|----|
| 15.   | Claim 15 -----                                                                                    | 61 |
| a.    | Preamble -----                                                                                    | 61 |
| b.    | First Virtual Microphone -----                                                                    | 61 |
| c.    | Second Virtual Microphone -----                                                                   | 61 |
| d.    | Substantially Similar Responses to Noise and<br>Substantially Dissimilar Responses to Speech----- | 62 |
| e.    | Virtual Microphone Array with a Single Null -----                                                 | 62 |
| f.    | Signal Processor -----                                                                            | 63 |
| g.    | Applying a Varying Linear Transfer Function-----                                                  | 63 |
| h.    | Generating an Output Signal with Reduced<br>Noise -----                                           | 64 |
| 16.   | Claim 16 -----                                                                                    | 65 |
| 17.   | Claim 17 -----                                                                                    | 65 |
| 18.   | Claim 18 -----                                                                                    | 66 |
| 19.   | Claim 19 -----                                                                                    | 66 |
| 20.   | Claim 20 -----                                                                                    | 67 |
| B.    | Ground 2: Brandstein, Gannot, and Griffiths-Jim -----                                             | 67 |
| C.    | Ground 3: Brandstein, Gannot, and McCowan -----                                                   | 69 |
| VII.  | SECONDARY CONSIDERATIONS OF<br>NONOBVIOUSNESS -----                                               | 72 |
| VIII. | DISCRETIONARY DENIAL UNDER §314(A) IS NOT<br>APPROPRIATE.-----                                    | 73 |
| A.    | Co-Pending Litigation Does Not Warrant Discretionary<br>Denial ( <i>Fintiv</i> )-----             | 73 |

*Meta Platforms, Inc. v. Jawbone Innovations, LLC*  
IPR Petition – U.S. Patent 11,122,357

|     |                                                                |    |
|-----|----------------------------------------------------------------|----|
| 1.  | Factor 1: Potential Stay -----                                 | 73 |
| 2.  | Factor 2: Proximity of Trial to FWD -----                      | 76 |
| 3.  | Factor 3: Investment in Parallel Proceeding -----              | 76 |
| 4.  | Factor 4: Overlapping Issues-----                              | 77 |
| 5.  | Factor 5: The Parties -----                                    | 77 |
| 6.  | Factor 6: Other Circumstances -----                            | 77 |
| B.  | Prior IPR Petitions ( <i>General Plastic</i> )-----            | 78 |
| IX. | DISCRETIONARY DENIAL UNDER §325(D) IS NOT<br>APPROPRIATE.----- | 80 |
| X.  | MANDATORY NOTICES-----                                         | 80 |
| A.  | Real Parties-In-Interest (37 C.F.R. §42.8(b)(1)) -----         | 80 |
| B.  | Related Matters (37 C.F.R. §42.8(b)(2))-----                   | 80 |
| C.  | Lead and Backup Counsel (37 C.F.R. §42.8(b)(3)) -----          | 81 |
| D.  | Service Information (37 C.F.R. §42.8(b)(4))-----               | 83 |
| E.  | Payment of Fees (37 C.F.R. §42.103) -----                      | 83 |
| F.  | Grounds for Standing (37 C.F.R. §42.104(a))-----               | 83 |
|     | CONCLUSION -----                                               | 83 |

## **TABLE OF AUTHORITIES**

|                                                                                                                                    | <b>Page(s)</b> |
|------------------------------------------------------------------------------------------------------------------------------------|----------------|
| <b>Cases</b>                                                                                                                       |                |
| <i>Apple v. Fintiv</i> ,<br>IPR2020-00019, Paper 11 (P.T.A.B. Mar. 20, 2020).....                                                  | 73, 77         |
| <i>Apple v. Uniloc 2017 LLC</i> ,<br>IPR2020-00854, Paper 9 (P.T.A.B. Oct. 28, 2020).....                                          | 78             |
| <i>Celltrion, Inc. v. Genentech, Inc.</i> ,<br>IPR2018-01019, Paper 11 (P.T.A.B. Oct. 30, 2018).....                               | 78             |
| <i>Central Security Group</i> ,<br>IPR2019-01609, Paper 11(P.T.A.B. Feb. 6, 2022).....                                             | 78             |
| <i>GAF Materials LLC v. Kirsch Research and Dev., LLC</i> ,<br>IPR2021-00192, Paper 14 (P.T.A.B. May 25, 2021) .....               | 76             |
| <i>General Plastic Indus. Co., Ltd. v. Canon Kabushiki Kaisha</i> ,<br>IPR2016-01357, Paper 19 (P.T.A.B. Sept. 6, 2017).....       | 78, 79         |
| <i>Global Tel*Link Corp. v. HLFIP Holding, Inc.</i> ,<br>IPR2021-00444, Paper 14 (P.T.A.B. Jul. 22, 2021) .....                    | 75             |
| <i>In re Google LLC</i> ,<br>58 F.4th 1379 (Fed. Cir. 2023) .....                                                                  | 73             |
| <i>Google LLC v. Jawbone Innovations, LLC</i> ,<br>IPR2022-00630, Paper 10 (P.T.A.B. Sept. 13, 2022).....                          | 77             |
| <i>In re GPAC Inc.</i> ,<br>57 F.3d 1573 (Fed. Cir. 1995) .....                                                                    | 23             |
| <i>Huawei Techs. Co. v. WSOU Invs., LLC</i> ,<br>IPR2021-00226, Paper 10 (P.T.A.B. June 10, 2021) .....                            | 76             |
| <i>Jawbone Innovations, LLC v. Amazon.com, Inc., et al.</i> ,<br>No. 2:21-cv-00435-JRG, Dkt. No. 60 (E.D. Tex Sept. 21, 2022)..... | 73, 81         |

|                                                                                                                                    |               |
|------------------------------------------------------------------------------------------------------------------------------------|---------------|
| <i>Meta Platforms, Inc. v. Jawbone Innovations, LLC</i><br>IPR Petition – U.S. Patent 11,122,357                                   |               |
| <i>Jawbone Innovations, LLC v. Meta Platforms, Inc.</i> ,<br>No. 6:23-cv-00158, Dkt. No. 1 (W.D. Tex. Feb. 28, 2023) .....         | 73            |
| <i>Jawbone Innovations, LLC v. Meta Platforms, Inc.</i> ,<br>No. 6:23-cv-00158-DC-DTG (W.D. Tex.).....                             | <i>passim</i> |
| <i>KSR Int’l Co. v. Teleflex Inc.</i> ,<br>550 U.S. 398 (2007).....                                                                | <i>passim</i> |
| <i>Leapfrog Enters. v. Fisher-Price, Inc.</i> ,<br>485 F.3d 1157 (Fed. Cir. 2007) .....                                            | 72            |
| <i>Mercedes-Benz USA, LLC v. Carucel Invs. L.P.</i> ,<br>IPR2019-01404, Paper 12 (P.T.A.B. Jan. 22, 2020) .....                    | 79            |
| <i>NetNut Ltd. v. Bright Data Ltd.</i> ,<br>IPR2021-00465, Paper 11 (P.T.A.B. Aug. 12, 2021).....                                  | 79            |
| <i>Newell Cos. v. Kenney Mfg. Co.</i> ,<br>864 F.2d 757 (Fed. Cir. 1988) .....                                                     | 72            |
| <i>Nidec Motor Corp. v. Zhongshan Broad Ocean Motor Co. Ltd.</i> ,<br>868 F.3d 1013 (Fed. Cir. 2017) .....                         | 24            |
| <i>Robert Bosch Healthcare Sys., Inc. v. Cardiocom, LLC</i> ,<br>No. C-14-1575 EMC, 2014 WL 3107447 (N.D. Cal. July 3, 2014) ..... | 74            |
| <i>Samsung Elecs. Am. Inc. v. Snik LLC</i> ,<br>IPR2020-01428, Paper 10 (P.T.A.B. Mar. 9, 2021).....                               | 76            |
| <i>Sand Revolution II, LLC v. Cont’l Intermodal Grp.-Trucking LLC</i> ,<br>IPR2019-01393, Paper 24 (P.T.A.B. June 16, 2020) .....  | 76            |
| <i>Skechers U.S.A., Inc. v. Nike, Inc.</i> ,<br>IPR2021-00160, Paper 10 (P.T.A.B. May 19, 2021) .....                              | 76            |
| <i>Symantec Corp. v. Zscaler, Inc.</i> ,<br>No. 17-cv-04426-JST, 2018 WL 3539267 (N.D. Cal. July 23,<br>2018) .....                | 74            |
| <i>Toshiba Am. Info. Sys., Inc. v. Wallelex Microelecs. Ltd.</i> ,<br>IPR2018-01538, Paper 11 (P.T.A.B. Mar. 5, 2019).....         | 79            |

*Meta Platforms, Inc. v. Jawbone Innovations, LLC*  
IPR Petition – U.S. Patent 11,122,357

|                                                                                                                    |    |
|--------------------------------------------------------------------------------------------------------------------|----|
| <i>Twilio, Inc. v. TeleSign Corp.</i> ,<br>No. 16-cv-06925-LHK, 2018 WL 1609630 (N.D. Cal. April 3,<br>2018) ..... | 74 |
|--------------------------------------------------------------------------------------------------------------------|----|

|                                                                                                                                |    |
|--------------------------------------------------------------------------------------------------------------------------------|----|
| <i>Uniloc USA Inc. v. LG Elecs., U.S.A. Inc.</i> ,<br>No. 18-cv-06737-JST, 2019 WL 1905161 (N.D. Cal. April 29,<br>2019) ..... | 74 |
|--------------------------------------------------------------------------------------------------------------------------------|----|

|                                                                                    |    |
|------------------------------------------------------------------------------------|----|
| <i>Unwired Planet, LLC v. Google Inc.</i> ,<br>841 F.3d 995 (Fed. Cir. 2016) ..... | 23 |
|------------------------------------------------------------------------------------|----|

|                                                                                                  |    |
|--------------------------------------------------------------------------------------------------|----|
| <i>Vivid Techs., Inc. v. Am. Sci. &amp; Eng’g, Inc.</i> ,<br>200 F.3d 795 (Fed. Cir. 1999) ..... | 25 |
|--------------------------------------------------------------------------------------------------|----|

**Statutes**

|                       |    |
|-----------------------|----|
| 35 U.S.C. § 112 ..... | 24 |
|-----------------------|----|

**Other Authorities**

|                             |    |
|-----------------------------|----|
| 37 C.F.R. §42.8(b)(1) ..... | 80 |
| 37 C.F.R. §42.8(b)(2) ..... | 80 |
| 37 C.F.R. §42.8(b)(3) ..... | 81 |
| 37 C.F.R. §42.8(b)(4) ..... | 83 |
| 37 C.F.R. § 42.10(b) .....  | 82 |
| 37 C.F.R. §42.15(a) .....   | 83 |
| 37 C.F.R. §42.103 .....     | 83 |
| 37 C.F.R. §42.104(a) .....  | 83 |

**TABLE OF EXHIBITS**

| <b>Exhibit No.</b> | <b>Description</b>                                                                                                                                                                                             |
|--------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| 1001               | U.S. Patent No. 11,122,357 (“the ’357 patent”)                                                                                                                                                                 |
| 1002               | Declaration of Richard M. Stern, Ph.D.                                                                                                                                                                         |
| 1003               | Excerpts of MICROPHONE ARRAYS: SIGNAL PROCESSING TECHNIQUES AND APPLICATIONS (Michael Brandstein & Darren Ward eds., Springer-Verlag 2001) (“Brandstein”)                                                      |
| 1004               | Sharon Gannot et al., <i>Signal Enhancement Using Beamforming and Nonstationarity with Applications to Speech</i> , vol. 49, no. 8 IEEE TRANSACTIONS ON SIGNAL PROCESSING, 1614 (Aug. 2001) (“Gannot”)         |
| 1005               | 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”) |
| 1006               | 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”)                                             |
| 1007               | U.S. Patent No. 5,651,071 (“Lindemann”)                                                                                                                                                                        |
| 1008               | U.S. Patent No. 5,627,799 (“Hoshuyama”)                                                                                                                                                                        |
| 1009               | U.S. Patent Publication No. 2003/0128848 (“Burnett ’848”)                                                                                                                                                      |
| 1010               | Excerpts from the ’357 patent’s file history                                                                                                                                                                   |
| 1011               | U.S. Provisional Patent Application No. 61/045,377                                                                                                                                                             |
| 1012               | Curriculum Vitae of Richard M. Stern, Ph.D.                                                                                                                                                                    |

*Meta Platforms, Inc. v. Jawbone Innovations, LLC*  
IPR Petition – U.S. Patent 11,122,357

| <b>Exhibit No.</b> | <b>Description</b>               |
|--------------------|----------------------------------|
| 1013               | Declaration of Carol S. Peterson |

Petitioner Meta Platforms, Inc. (“Petitioner” or “Meta”), requests *inter partes* review of claims 1-20 (“Challenged Claims”) of U.S. Patent 11,122,357 (“the ’357 patent”), which Jawbone Innovations (“Patent Owner” or “PO”) purportedly owns. This petition, accompanied by a motion for joinder, presents the same grounds as IPR2023-00251 (the “Amazon IPR”), which was instituted on June 1, 2023.

## INTRODUCTION

The Challenged Claims recite devices that process audio signals from microphones to reduce noise. The claimed devices comprise two “virtual” microphones, each formed by combining signals from two physical microphones. The two virtual microphones must have substantially similar responses to noise and substantially dissimilar responses to speech. But the claims do not require that the device do anything with the virtual microphones. Instead, the claims merely recite that the device must include a signal processor that performs generic signal-processing operations like filtering the *physical*-microphone signals, summing the *physical*-microphone signals, and applying a transfer function. The claims recite these conventional signal-processing operations only at a high level. For example, the claims do not elaborate on the filter to apply; they require only that some “filtering” of the physical-microphone signals occur.

The two virtual microphones may be created by the recited filtering and summing of the physical-microphone signals. But as the ’357 patent concedes, that was

a “common” technique for creating virtual microphones known to those skilled in the art. (Ex. 1001, 8:55-60.) The generic language of the claims contrasts with the ’357 patent specification, which identifies specific formulas defining the two virtual microphones that are the basis for the purported innovation. (*Compare, e.g., id.*, claim 1 *with id.*, 11:6-16, 12:20.)

Untethered from the specification’s formulas, the claims’ recitation of generic signal-processing concepts encompasses prior art describing the Generalized Side-lobe Canceler (“GSC”), a fundamental noise-reduction technique introduced in the 1980s. Broadly applicable to many signal-processing applications, the GSC involves filtering and summing the signals from at least two sensors in different ways to produce two virtual sensors, one that captures the target signal plus noise and another that captures just noise. Subtracting the noise signal from the target-plus-noise signal cancels out the noise and yields a cleaner output signal.

Years before the ’357 patent’s earliest possible priority date, a widely used reference book, MICROPHONE ARRAYS (Springer-Verlag 2001) (“Brandstein”), explained that it was common to use the GSC with a microphone array to reduce noise in speech-signal processing. Contemporaneously with Brandstein’s publication, Sharon Gannot and other researchers published in IEEE’s Transactions on Signal Processing an article titled *Signal Enhancement Using Beamforming and Nonstationarity with Applications to Speech* (“Gannot”), describing a generalized version

of the GSC technique that would make it even more robust by handling arbitrary linear transfer functions. As its title indicates, Gannot likewise contemplated reducing noise in speech applications. Together, Brandstein and Gannot disclose all the limitations of the '357 patent's claims and render all the claims obvious.

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

## **I. BACKGROUND**

When Patent Owner filed the '357 patent's priority applications in 2007, techniques for reducing noise in signals had been known for decades. One prominent technique was the Generalized Sidelobe Canceler, or GSC. In a nod to its inventors, the GSC is sometimes also called the Griffiths-Jim beamformer. (Ex. 1002 ¶30.)

### **A. Griffiths and Jim Publish Their Seminal GSC Article in 1982.**

In 1982, Lloyd Griffiths and Charles Jim published a paper describing “a simple time-varying beamformer which can be used to combine the outputs of an array of sensors.” (Ex. 1005, 27.) The beamformer's purpose was “to minimize the effects of noise and interference at the array output” while capturing the target signal. (*Id.*)

Griffiths and Jim called their beamformer a “generalized sidelobe canceling” structure. (*Id.*, 29.) Illustrated in Figure 4 of their paper, the signal processor 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 captures only noise so that the noise could be subtracted or canceled out:

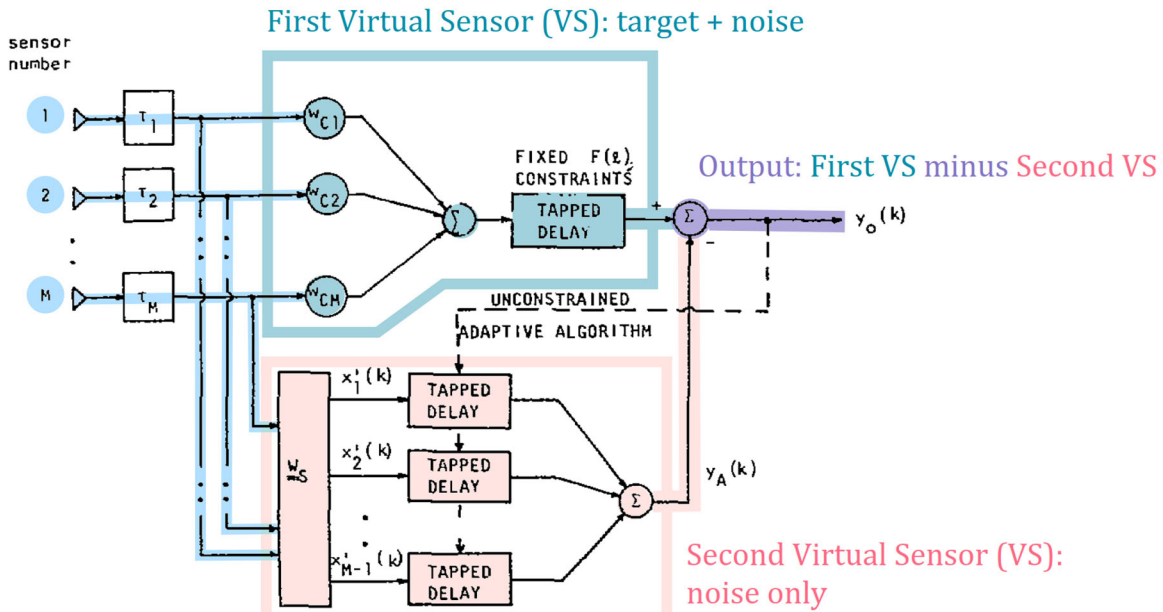


Fig. 4. Generalized sidelobe canceling form of linearly constrained adaptive array processing algorithm.

(*Id.*, 29–30.<sup>1</sup>) In the top branch, the outputs of the **array sensors** were combined to form a conventional beamformer, which Petitioner calls the **first virtual sensor**. The sensor outputs were combined by multiplying the sensor output signals by factors called “weights” ( $w_{C1}, \dots, w_{CM}$  in the paper, and sometimes also called “gains”) and further filtering and summing the weighted sensor signals. (*Id.*; Ex. 1002 ¶35.)

<sup>1</sup> Figures have been annotated with color throughout.

In the paper’s equations, the output of the first virtual sensor was denoted  $y_c'(k)$ . (Ex. 1005, 30.) This output contains the target signal plus noise. (Ex. 1002 ¶35.)

“The lower path in Fig. 4 is the sidelobe canceling path” (Ex. 1005, 30), which Petitioner calls the **second virtual sensor**. Like the first virtual sensor, the second virtual sensor is formed from a combination of the outputs of the array sensors, but the combination differs from the combination used for the first virtual sensor. The lower path includes a blocking matrix designed to “block the desired signal  $s(k)$  from the lower path.” (*Id.*) The array sensor outputs were combined by filtering and summing (including by blocking the desired signal), and the output of the second virtual sensor was denoted in the paper as  $y_A(k)$ . (*See id.*) The second virtual sensor’s output “ $y_A(k)$  contains no desired signal terms” and instead “contains only noise and interference terms.” (*Id.*)

The overall output of the GSC,  $y(k)$ , was produced by subtracting the noise-only output of the second virtual sensor from the target-plus-noise output of the first virtual sensor:

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

(*Id.*) The result was a cleaned-up signal that reduced noise without distorting the desired signal. (Ex. 1002 ¶40; Ex. 1005, 30 (output due to desired signal satisfies

the constraint defined by paper’s equation 9, which defines a constraint for “zero distortion” (p. 28)).)

**B. Over the Next Two Decades, the GSC was Used in Microphone Arrays to Reduce Noise.**

In the twenty years following Griffiths and Jim’s article, the GSC was used in many signal-processing applications, including with microphone arrays to reduce noise in speech applications. For example, U.S. Patent No. 5,651,071 (“Lindemann”), filed in 1993, cites the article and explains that using a Griffiths-Jim beamformer “to improve signal-to-noise ratio for hearing aids” was known. (Ex. 1007, 1:40-46, 12:12–14.<sup>2</sup>) As another example, Griffiths-Jim is the first non-patent reference cited in U.S. Patent No. 5,627,799 (“Hoshuyama”), filed in 1995, which relates to “interference cancelers, and more particularly to a generalized sidelobe canceler, or adaptive beamformer for an array of sensors such as microphones[.]” (Ex. 1008, 1:8-11.) Describing what was prior art even then, Hoshuyama explains one way the GSC had been used with microphone arrays:

According to a prior art microphone array, signals detected by an array of microphones are lowpass filtered and summed together to detect a target signal that arrives in a particular direction. The adaptive microphone array beamformer is one form of the generalized sidelobe canceler as described in an article “An alternative Approach to Linearly Constrained Adaptive Beamforming”, Lloyd J. Griffiths and Charles

---

<sup>2</sup> Patent citations are in column:line format.

W. Jim, the IEEE Transactions on Antenna and Propagation, Vol. AP-30, No. 1, January 1982, pages 27-34.

(*Id.*, 1:17-26.)

**C. Published in 2001, Brandstein Illustrates How to Use a GSC with a Microphone Array to Reduce Noise.**

In 2001, MICROPHONE ARRAYS: SIGNAL PROCESSING TECHNIQUES AND APPLICATIONS published. (Ex. 1003 (“Brandstein”).) The editors’ goal was to provide “a single complete reference on microphone arrays.” (*Id.*, Preface.) The book quickly became a standard reference for those in the field of audio-signal processing. (Ex. 1002 ¶45.)

At the outset of the chapter on robust adaptive beamforming, Brandstein explains that “[a]pplications of beamforming include microphone arrays for speech enhancement.” (Ex. 1003, 87 (original page numbering).) “The goal of speech enhancement is to remove undesirable signals such as noise and reverberation.” (*Id.*) Brandstein further explains that, among various known adaptive beamformers, “the Griffiths-Jim beamformer (GJBF), or the generalized sidelobe canceler, is most widely known.” (*Id.*, 88 (internal citation omitted).) “Figure 5.1 depicts the structure of the GJBF.” (*Id.*)

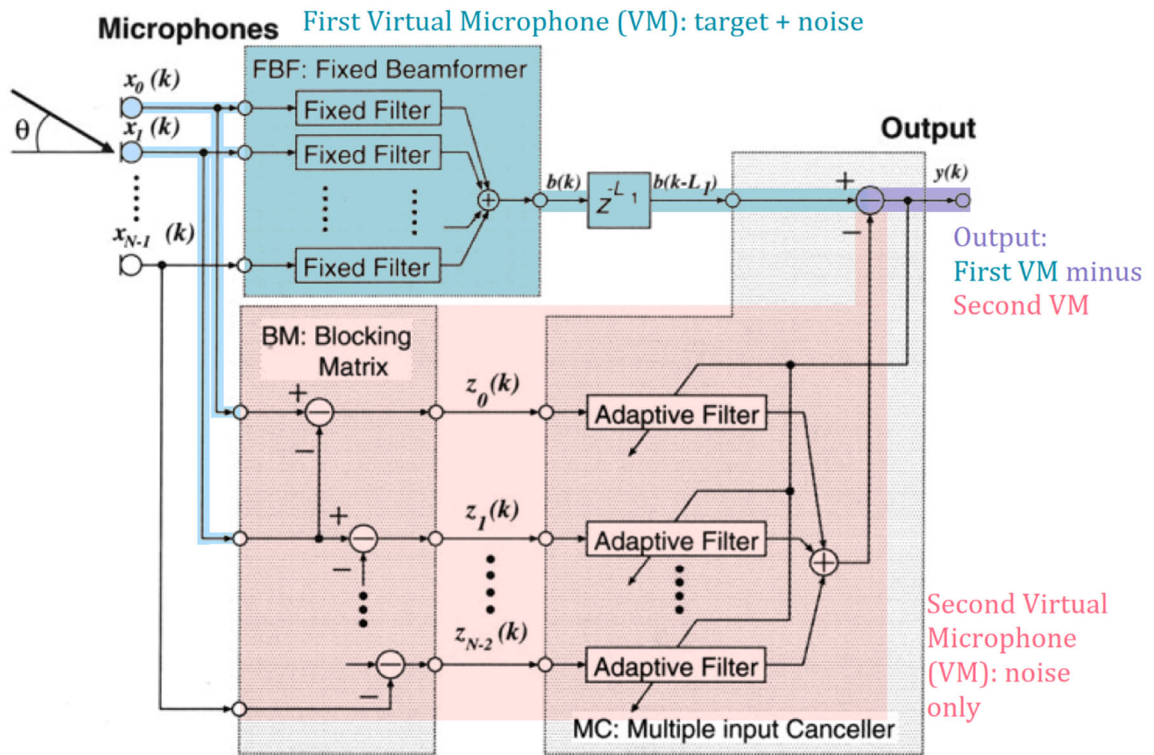


**Fig. 5.1.** Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceler (MC), and a blocking matrix (BM).

(*Id.*, 89, Fig. 5.1.) As shown, the signals of at least **two physical microphones**,  $x_0(k)$  and  $x_1(k)$ , are combined in the top branch by filtering and summing the signals to form a fixed beamformer—a **first virtual microphone**. (Ex. 1002 ¶48.) The first virtual microphone captures the target speech signal plus noise. (*Id.*)

In the bottom branch, the signals of the **two physical microphones** are combined by filtering and summing the signals in a different way to form a **second virtual microphone**. (*Id.* ¶49.) The second virtual microphone includes a blocking matrix (BM). (*Id.*) “[T]he BM forms a null in the look direction so that the target signal is suppressed and all other signals are passed through.” (Ex. 1003, 88.) “The

BM was named after its function, which is to block the target signal.” (*Id.*) As a result, the second virtual microphone captures noise only. (Ex. 1002 ¶49.)

The overall **output** is the target-plus-noise output of the first virtual microphone minus the noise-only output of the second virtual microphone. (*Id.* ¶50.) The result is that, “in the subtracter output  $y(k)$ , the target signal is enhanced and undesirable signals such as ambient noise and interferences are suppressed.” (Ex. 1003, 88-89.)

The two virtual microphones have very different responses to the target speech signal: the first virtual microphone is designed to capture the target signal, while the second virtual microphone is designed to block it. (Ex. 1002 ¶51.) On the other hand, they have similar responses to noise so that in the final subtraction output the noise is removed. (*Id.*) This is illustrated in Figure 5.2, which shows the directivity pattern for the final output of an example Griffiths-Jim beamformer:

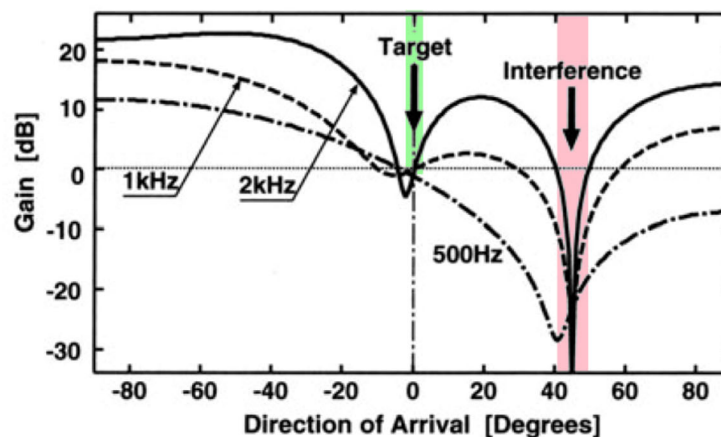


**Fig. 5.2.** Example directivity pattern of the Griffiths-Jim beamformer.

(Ex. 1003, 89, Fig. 5.2.) The horizontal axis of the graph shows direction of arrival

measured in degrees relative to the microphone array: the target signal is shown at 0 degrees, while the noise signal is shown at approximately 45 degrees. (Ex. 1002 ¶51.) The vertical axis of the graph shows 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 green, the target signal is reproduced faithfully with essentially zero gain, reflecting that subtracting the second virtual microphone's noise-only output from the first virtual microphone's target-plus-noise output will produce the target signal in the direction of the target. (*Id.* ¶52.) On the other hand, as highlighted in red, Figure 5.2 shows a highly negative gain in the direction of the interference signal, reflecting that subtracting the two virtual microphones' outputs cancels the noise. (*Id.* ¶53; *see also* Ex. 1003, 90 (“In the direction of the target signal, almost constant gains close to 0 dB are obtained over a wide range of frequencies. On the contrary, in the direction of the interference, a deep null is formed.”).)

**D. Contemporaneously, Gannot Taught Adapting the GSC to Handle Arbitrary Transfer Functions.**

In August 2001, IEEE's Transactions on Signal Processing publication (vol. 49, no. 8) included an article titled *Signal Enhancement Using Beamforming and Nonstationarity with Applications to Speech* by Sharon Gannot, David

Burshtein, and Ehud Weinstein. (Ex. 1004 (“Gannot”).) As its title indicates, the article contemplates speech enhancement through beamforming. (*Id.*) Specifically, the article considers a sensor array “where arbitrary transfer functions (TFs) relate the source signal and the sensors.” (*Id.*, 1614 (Abstract).) As an audio signal travels from its source to a microphone, the signal may change, such that the signal received at the microphone is not exactly the same as the signal when it originated from the source. (Ex. 1002 ¶55.) The acoustic path from the source to the microphone can be thought of as a system that brings about this change, and the operation of the acoustic path on the signal can be represented mathematically by a transfer function. (*Id.*)

Gannot notes that the generalized sidelobe canceler (GSC) works well when the acoustic paths’ transfer functions satisfy certain criteria, such as when the signals received at the sensors “are simple delayed versions of the source signal.” (Ex. 1004, 1614 (Abstract).) But the original Griffiths-Jim GSC may suppress interference poorly “in complicated acoustic environments, where arbitrary TFs [transfer functions] may be encountered.” (*Id.*) Gannot thus proposes a GSC solution adapted to handle arbitrary transfer functions. (*Id.*)

Gannot's Figure 3 shows the proposed GSC structure, and Figure 4 summarizes the algorithm:

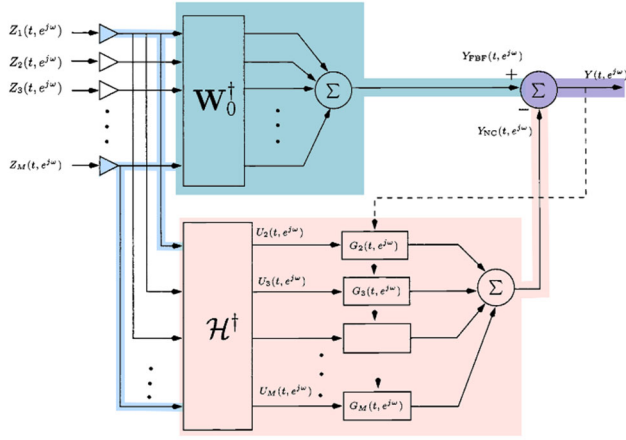


Fig. 3. Linearly constrained adaptive beamformer.

- 1) TF-s ratios:  $\mathbf{H}(e^{j\omega}) = \frac{\mathbf{A}(e^{j\omega})}{A_1(e^{j\omega})}$
  - 2) Fixed beamformer:  
 $Y_{\text{FBF}}(t, e^{j\omega}) = \mathbf{W}_0^\dagger(e^{j\omega})\mathbf{Z}(t, e^{j\omega})$
  - 3) Noise reference signals:  
 $\mathbf{U}(t, e^{j\omega}) = \mathcal{H}^\dagger(e^{j\omega})\mathbf{Z}(t, e^{j\omega})$
  - 4) Output signal:  
 $Y(t, e^{j\omega}) = Y_{\text{FBF}}(t, e^{j\omega}) - \mathbf{G}^\dagger(t, e^{j\omega})\mathbf{U}(t, e^{j\omega})$
  - 5) Filters update, for  $m = 1, \dots, M-1$ :  
 $\tilde{G}_m(t+1, e^{j\omega}) = G_m(t, e^{j\omega}) + \mu \frac{U_m(t, e^{j\omega})Y^*(t, e^{j\omega})}{P_{\text{est}}(t, e^{j\omega})}$   
 $G_m(t+1, e^{j\omega}) \stackrel{\text{FIR}}{\leftarrow} \tilde{G}_m(t+1, e^{j\omega})$
- where,  
 $P_{\text{est}}(t, e^{j\omega}) = \rho P_{\text{est}}(t-1, e^{j\omega}) + (1-\rho) \sum_m |Z_m(t, e^{j\omega})|^2$   
 6) keep only non-aliased samples.  
 (note:  $\mathbf{W}_0(e^{j\omega})$  is defined in (16).  
 $\mathcal{H}(e^{j\omega})$  is defined in (19)).

Fig. 4. Suggested algorithm.

Ex. 1004, Fig. 3

Ex. 1004, Fig. 4

The blocking matrix  $\mathcal{H}^\dagger$  is used to create noise reference signals  $U_m$  that apply the linear transfer functions  $A_m$  of the acoustical paths:

$$U_m(e^{j\omega}) = Z_m(t, e^{j\omega}) - \frac{A_m(e^{j\omega})}{A_1(e^{j\omega})} Z_1(t, e^{j\omega})$$

$$m = 2, \dots, M. \quad (20)$$

(Ex. 1004, 1618-20.) By incorporating these terms, Gannot's more-general GSC accounts for arbitrary transfer functions. In particular, by using the ratio of the transfer function for microphone  $m$  to the transfer function for microphone 1, Gannot's

algorithm applies a transfer function for an acoustical path between the two microphones.<sup>3</sup> (Ex. 1002 ¶61.)

**E. Other Concepts in the '357 Patent Were Well Known.**

**1. Filtering and Summing in the Time Domain Were Well Known.**

In the microphone-array context, the GSC involves filtering and summing physical-microphone signals to create virtual microphones. It has been known since the outset that the GSC can be implemented in the time domain, a term that refers to analyzing signals as a function of time. (Ex. 1002 ¶35.) Both the original Griffiths-Jim paper and Brandstein show the sensor signals as functions of time. (*Id.* ¶¶35, 48.) Indeed, Lindemann discloses that using the GSC in a microphone array for hearing aids was a “time domain approach.” (Ex. 1007, 1:51-52.)

---

<sup>3</sup> The inventor named on the '357 patent had another application publish as U.S. 2003/0128848 (“Burnett '848”), which is prior art to the '357 patent. (Ex. 1009.) Burnett '848 also discloses using a ratio of transfer functions representing the transfer function of an acoustical path between microphones. (Ex. 1002 ¶¶71-75.) The '357 patent's discussion of transfer functions essentially repeats Burnett '848's disclosure, which confirms that this feature was known in the art. (*Id.* ¶83.)

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

Adaptive beamforming relies on the spatial geometry of the array and the signal's direction of arrival. (Ex. 1002 ¶¶62; Ex. 1003, 87.) When the source of the target signal is close enough to the array, instead of treating the signal's wavefront as a plane, it may be useful to account for the spherical geometry of the wavefront—i.e., to use a *near-field* design. (See Ex. 1002 ¶¶63-64.) Near-field designs were well known before the '357 patent and use basic calculations to determine the delays between signal arrival at the various microphones in the array. (*Id.*) Brandstein discloses the standard near-field design, using  $c$  to denote the speed of sound:

- 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.

(Ex. 1003, 25.) The time it takes for a signal to travel from a source to a microphone is the distance traveled divided by the speed of sound. (Ex. 1002 ¶¶65.) The  $\tau_i$  terms above thus provide the difference in time it takes for the signal to propagate from

the source to microphone  $i$  relative to a reference microphone. (*Id.*) Including the  $\tau_i$  term in the complex exponential shown above and applying that exponential to the signal received at the reference microphone 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 the January 2002 edition of the journal DIGITAL SIGNAL PROCESSING, in an article by McCowan et al. titled *Near-Field Adaptive Beamformer for Robust Speech Recognition* (“McCowan”). (Ex. 1006, 89-90.) McCowan expressly motivates and teaches incorporating the near-field design into the GSC. (*Id.*, 90-96.)

## II. THE '357 PATENT

### A. The '357 Patent Discloses Nothing Innovative.

#### 1. The '357 Patent Purports to Distinguish Itself from the Prior Art by Using a Virtual Microphone Designed to Capture Only Noise, Which Had Been Known for Decades.

The '357 patent asserts that, “conventional arrays and algorithms ... seek to reduce noise by nulling out noise sources,” whereas the '357 patent uses a virtual microphone with a null in the direction of speech “to remove the speech of the user.” (Ex. 1001, 5:10-16.) But using a virtual microphone with a null in the direction of the target signal to obtain a noise-only reference is exactly what the GSC does. (Ex. 1002 ¶¶39, 49.)

#### 2. The '357 Patent Concedes It Relies on Known Techniques to Form Virtual Microphones from Physical Microphones.

The '357 patent is directed to forming virtual microphones from a dual omnidirectional microphone array. Figure 2 illustrates the array:

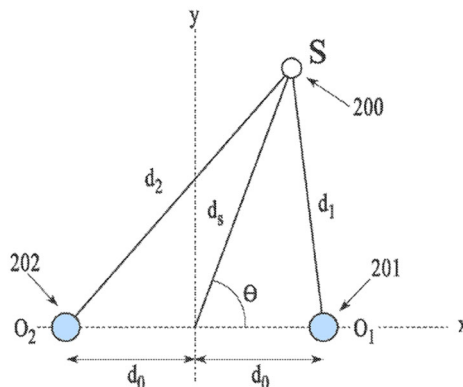


FIG.2

(Ex. 1001, Fig. 2.) There are two physical microphones  $O_1$  and  $O_2$  (201 and 202).

(Ex. 1001, 8:40-42.) The source  $S$  (200) of the target signal is also shown. (*Id.*)

Figure 3 illustrates combining the two physical microphone signals to create a virtual microphone:

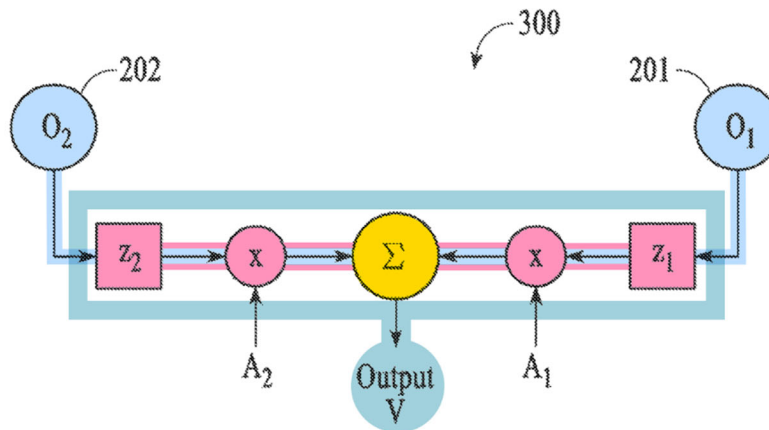


FIG.3

(*Id.*, Fig. 3.) “The **output from each microphone** 201 and 202 can be **delayed ( $Z_1$  and  $Z_2$ )**, **multiplied by a gain ( $A_1$  and  $A_2$ )**, and then **summed** with the other as demonstrated in FIG. 3.” (*Id.*, 8:50-53.) “The output of the array is or forms at least one virtual microphone ....” (*Id.*, 8:53-54.)

The '357 patent acknowledges this was well known in the art: “By varying the magnitude and sign of the delays and gains, a wide variety of virtual microphones (VMs), also referred to herein as virtual directional microphones, can be realized. *There are other methods known to those skilled in the art for constructing VMs but*

*this is a common one and will be used in the enablement below.” (Id., 8:55-61 (emphasis added).)*

Figure 4 illustrates the same concept with two processing paths to form two virtual microphones,  $V_1$  and  $V_2$ :

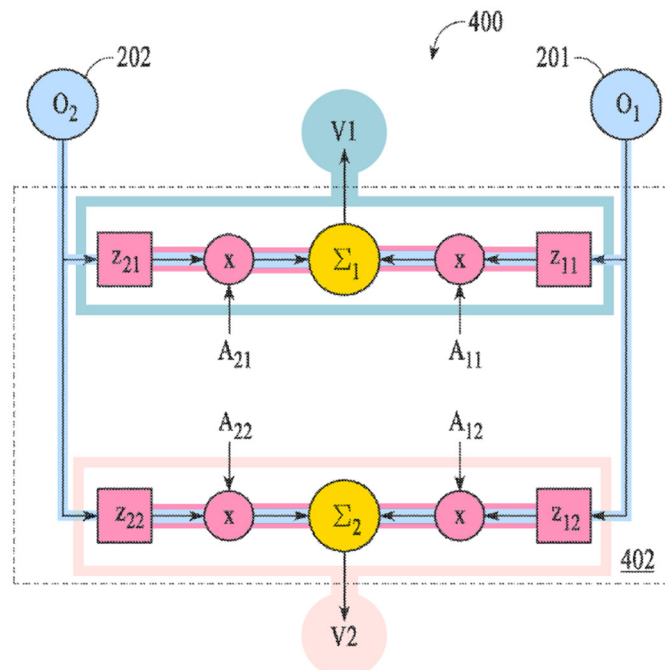


FIG.4

(Id., Fig. 4.)

### 3. The '357 Patent Discloses Formulas for the Purportedly Innovative Set of Virtual Microphones, But These Formulas Rely on Near-Field Design.

The '357 patent discloses using two virtual microphones,  $V_1$  and  $V_2$ , defined by the following formulas:

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

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

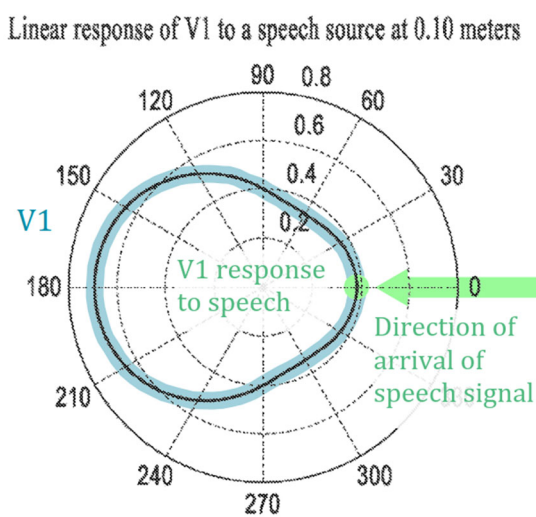
(Ex. 1001, 12:20, 11:8). In these formulas,  $O_1$  and  $O_2$  are the outputs of the two physical microphones, and:

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

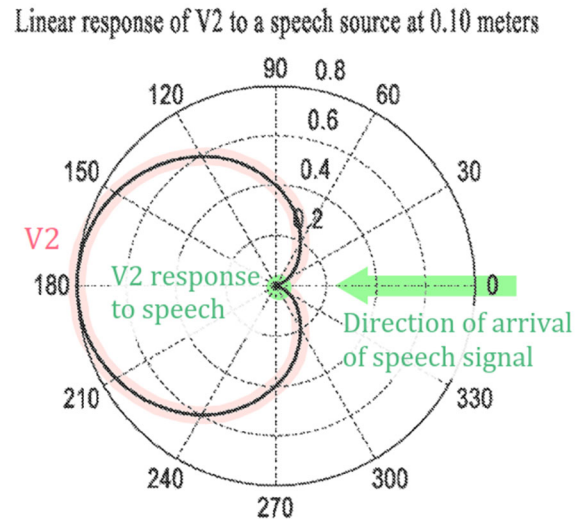
(*Id.*, 11:10-20.) “The distances  $d_1$  and  $d_2$  are the distance from  $O_1$  and  $O_2$  to the speech source (see FIG. 2), respectively, and  $\gamma$  is their difference divided by  $c$ , the speed of sound, and multiplied by the sampling frequency  $f_s$ .” (*Id.*, 11:19-22.) Thus,  $\gamma$  is simply the difference in propagation time to the microphones taken from the standard near-field design equations (denoted  $\tau_i$  in §I.E.2, *supra*), just expressed in number of samples instead of other units of time, like seconds. Similarly,  $\beta$  is simply the amplitude-adjustment factor from the standard near-field design (denoted  $a_i$  in §I.E.2, *supra*). (Ex. 1002 ¶80.)

According to the ’357 patent, using these specific formulas for  $V_1$  and  $V_2$  produces desirable directivity patterns with respect to a speech signal expected to arrive

at an angle of zero degrees from 10 centimeters away, as illustrated in Figures 9 and 11:



*Ex. 1001, Fig. 11*



*Ex. 1001, Fig. 9*

As shown,  $V_1$  exhibits a response to the speech signal, while  $V_2$  has a null in the expected direction of the speech signal “and will therefore exhibit minimal response to the speech.” (*Id.*, 11:50-52, 12:39-43.) In contrast, the ’357 patent explains that  $V_1$  and  $V_2$  will supposedly have “very similar” responses to noise originating from a meter away from the array. (*Id.*, 12:44-48.)

## **B. The Claims Recite Generic Virtual Microphones and Generic Signal Processing.**

Independent claims 1 and 15 do not recite the formulas for  $V_1$  and  $V_2$  set forth in the specification. Instead, claims 1 and 15 recite simply combining the signals from two physical microphones in different ways to create two virtual microphones. The only other limitation that claim 1 imposes on the virtual microphones is that

they have substantially similar responses to noise and substantially dissimilar responses to speech. Claim 15 requires the same and adds only that a virtual microphone array including the two virtual microphones have only one null oriented toward a speech source.

Apart from reciting the virtual microphones, claims 1 and 15 merely recite that the device must include a signal processor that performs the generic operation of combining the signals of the *physical* microphones by filtering and summing in the time domain. The signal processor is also operative to apply a varying linear transfer function between the microphone signals. The claims neither specify any particular filter nor further limit the transfer function. Finally, claims 1 and 15 recite the end result: the signal processor must be operative to “generate an output signal having noise content that is attenuated with respect to speech content.”

Dependent claims add unpatentable details, such as that the signal processor comprises a digital signal processor or that the signal processor is operative to add a delay to a microphone signal.

**C. The Claims Were Not Carefully Scrutinized During Prosecution.**

The PTO Examiner did not consider the references Petitioner relies on. In fact, despite the generic nature of the claims and wealth of available prior art, the PTO Examiner did not cite *any* prior art against the issued claims.

The '357 patent issued from an application with only one original claim. (Ex. 1010, 184.) After the Examiner rejected that claim based on prior art, the applicant canceled that claim and added the twenty claims that issued. (*Id.*, 111-116.) The Examiner issued no anticipation or obviousness rejections against those claims (*see generally id.*, 1-108) and provided no substantive reasons for allowance (*id.*, 8, 48, 64).

### **III. STATEMENT OF RELIEF REQUESTED**

#### **A. Grounds**

The Board should cancel the claims as follows:

| Ground | Claims Challenged       | References                        |
|--------|-------------------------|-----------------------------------|
| 1      | 1-20                    | Brandstein, Gannot                |
| 2      | 4, 13, 14, 18, 19       | Brandstein, Gannot, Griffiths-Jim |
| 3      | 4, 5, 6, 13, 14, 18, 19 | Brandstein, Gannot, McCowan       |

#### **B. The Earliest Priority Date the '357 Patent Claims Is June 13, 2007.**

The earliest application to which the '357 patent claims priority is U.S. provisional application 60/934,551. Accordingly, the earliest possible priority date for all Challenged Claims is the '551 application's filing date, June 13, 2007. Petitioner does not concede that the claims are entitled to that priority date.

**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 (Ex. 1003; Ex. 1013 ¶¶19-30; Ex. 1002 ¶44);
- (2) Gannot published no later than 2001 (Ex. 1004; Ex. 1013 ¶¶31-42);
- (3) Griffiths-Jim published no later than 1982 (Ex. 1005; Ex. 1013 ¶¶43-54);  
and
- (4) McCowan published no later than 2002 (Ex. 1006; Ex. 1013 ¶¶55-66).

**D. The Asserted References Are Analogous Art.**

Prior art is analogous if (1) it is from the same field of endeavor, regardless of the problem addressed, or (2) 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 art because they are from the same field as the '357 patent, e.g., microphone array signal processing and noise reduction. (Ex. 1002 ¶20; Ex. 1001, 1:23-26, 5:8-9.) They are also pertinent to a problem the inventor was focused on, e.g., noise suppression. (Ex. 1002 ¶20; Ex. 1001, 1:23-26, 5:8-9.)

#### **IV. 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. (Ex. 1002 ¶¶27-28.) Work experience could substitute for formal education, and additional formal education could substitute for work experience. (*Id.*)

#### **V. CLAIM CONSTRUCTION**

On February 28, 2023 Patent Owner sued Petitioner for infringement of the ’357 patent in *Jawbone Innovations, LLC v. Meta Platforms, Inc.*, No. 6:23-cv-00158-DC-DTG (W.D. Tex.) (“the Litigation”). Although the parties may propose constructions for certain terms in the Litigation for purposes of determining infringement, no claim terms require construction to resolve the obviousness challenges here.<sup>4</sup> *Nidec Motor Corp. v. Zhongshan Broad Ocean Motor Co. Ltd.*, 868 F.3d

---

<sup>4</sup> In the Litigation, the parties may dispute whether some claims are invalid under 35 U.S.C. § 112. For this proceeding only, Petitioner assumes the claims are not invalid under § 112.

1013, 1017 (Fed. Cir. 2017); *Vivid Techs., Inc. v. Am. Sci. & Eng'g, Inc.*, 200 F.3d 795, 803 (Fed. Cir. 1999).

## **VI. GROUNDS OF UNPATENTABILITY**

### **A. Ground 1: Brandstein and Gannot**

Claims 1-20 would have been obvious in view of Brandstein and Gannot.

#### **1. Claim 1**

##### **a. Preamble**

The preamble recites: “A device, comprising ....” Brandstein discloses a device with a microphone array and signal processing that implements the GSC. (Ex. 1003, 89, Fig. 5.1.) It further discloses the use of beamforming in devices such as telephones and hearing aids. (*Id.*, 87.) Brandstein also expressly reports the results of experiments performed with a real-world implementation of a GSC on a system illustrated in Figure 5.16:

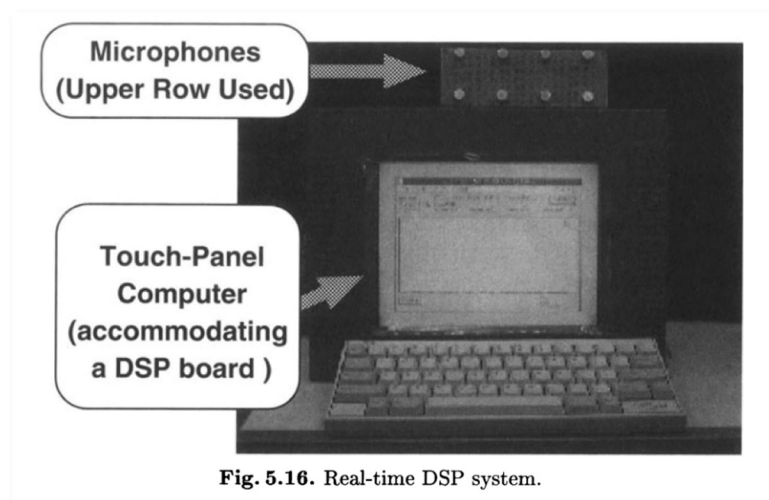


**Fig. 5.16.** Real-time DSP system.

(*Id.*, 105, Fig. 5.16 (annotations in original).) “The system comprises a microphone array and a compact touch-panel personal computer which includes a floating point DSP, the ADSP-21062.” (*Id.*, 104 (internal citation omitted).) Thus, Brandstein discloses a device. Alternatively, a POSITA would have understood that Brandstein’s GSC shown in Figure 5.1 would have been implemented in a device such as a hearing aid or telephone (as Brandstein teaches), and therefore it would have been obvious. (Ex. 1002 ¶85.)

**b. First Virtual Microphone Comprising a Combination of Signals from First and Second Physical Microphones**

Claim element 1[a] recites “a first virtual microphone comprising 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.

Figure 5.1 shows **first and second physical microphones** generating **first and second microphone signals** (e.g.,  $x_0$  and  $x_1$ ) and a **first virtual microphone** along the top branch of the GSC. The first virtual microphone comprises a first combination of the microphone signals:

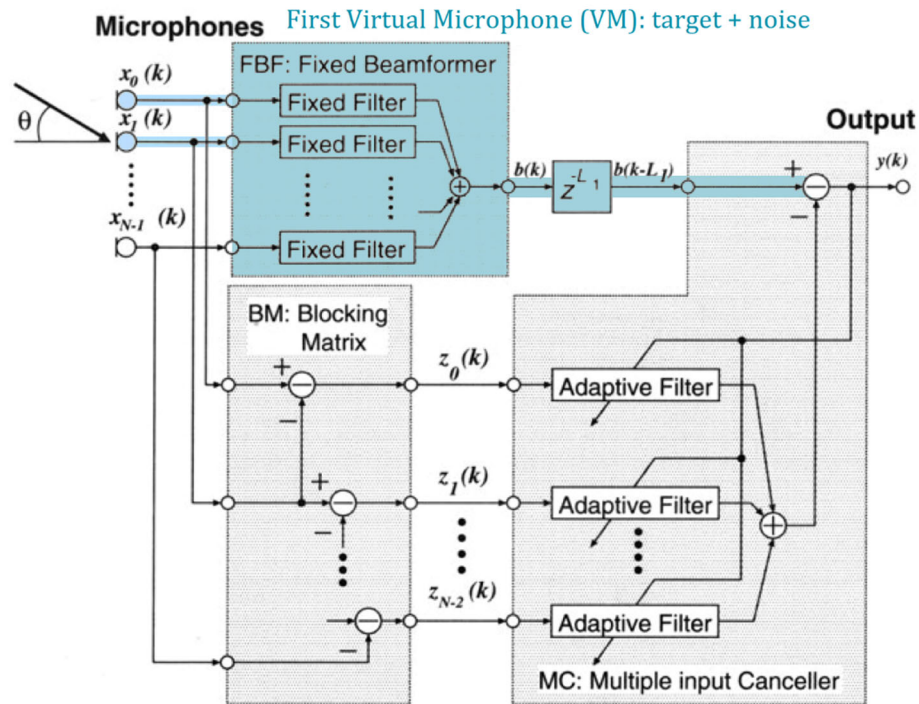


**Fig. 5.1.** Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceller (MC), and a blocking matrix (BM).

(Ex. 1003, 89, Fig. 5.1; Ex. 1002 ¶89.) For example, Brandstein discloses that the physical microphones may be omnidirectional microphones. (Ex. 1003, 101; Ex. 1002 ¶89.)

### c. Second Virtual Microphone

Claim element 1[b] begins by reciting: “a second virtual microphone comprising 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.

Figure 5.1 shows a **second virtual microphone** along the bottom branch of the GSC comprising a different combination of the **first and second microphone signals** (e.g.,  $x_0$  and  $x_1$ ):

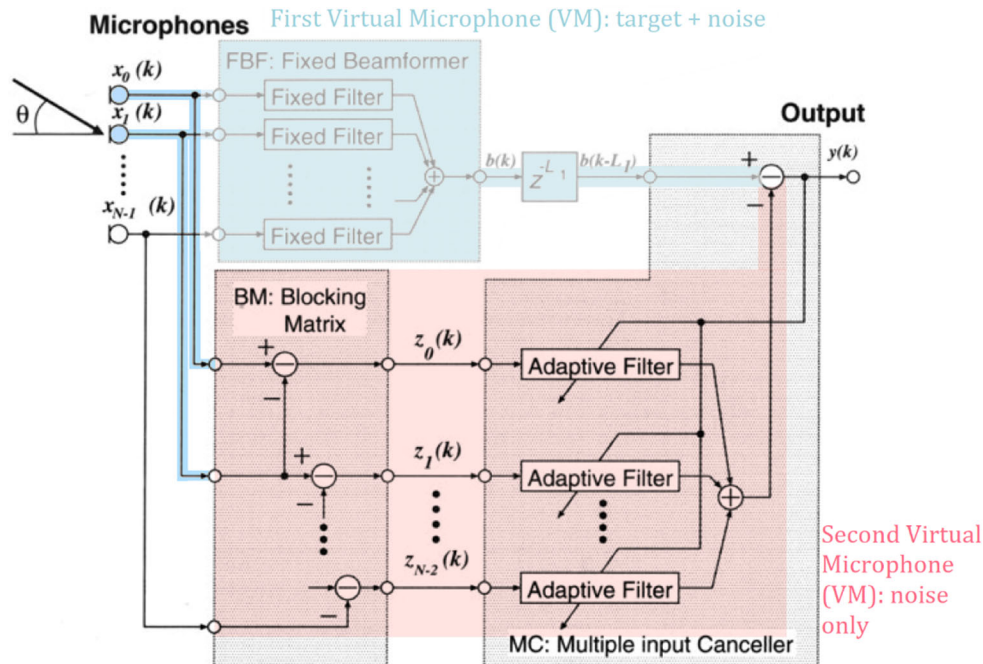


**Fig. 5.1.** Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceler (MC), and a blocking matrix (BM).

(Ex. 1003, 89, Fig. 5.1; Ex. 1002 ¶91.)

**d. Substantially Similar Responses to Noise and Substantially Dissimilar Responses to Speech**

Claim element 1[b] further recites: “wherein the first virtual microphone and the second virtual microphone are distinct virtual directional microphones with substantially similar responses to noise and substantially dissimilar responses to speech.” Brandstein discloses this limitation.

The virtual microphone formed in the top branch of Brandstein’s Figure 5.1 is distinct from the virtual microphone formed in the bottom branch. (If they were the same, the output of the GSC would be zero.) (Ex. 1002 ¶94.)

The two virtual microphones have very different responses to the target signal: the first virtual microphone is designed to form a beam that captures the target signal, while the second virtual microphone includes a blocking matrix designed to block the target signal. (*Id.*; Ex. 1003, 88.) Brandstein discloses that the target signal can be speech. (*E.g.*, Ex. 1003, 101 (“The target source was an English male speech signal.”); *also id.*, 103, Figs. 5.13, 5.14; Ex. 1002 ¶94.) Indeed, just before noting that the GSC was the most widely known adaptive beamformer, Brandstein states: “Applications of beamforming include microphone arrays for speech enhancement. The goal of speech enhancement is to remove undesirable signals such as noise and reverberation.” (Ex. 1003, 87.)

The two virtual microphones also have similar responses to noise so that in the final subtraction output the noise cancels out. (Ex. 1002 ¶95.) The different responses to the target signal and similar responses to noise are evident from Figure 5.2, which shows the directivity pattern for the final output of an example Griffiths-Jim beamformer:

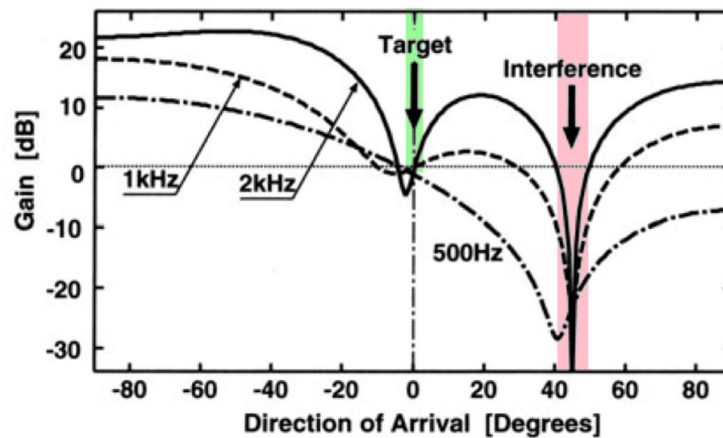


**Fig. 5.2.** Example directivity pattern of the Griffiths-Jim beamformer.

(Ex. 1003, 89, Fig. 5.2; Ex. 1002 ¶95.)

As highlighted in green, in the direction of the target signal, the Griffiths-Jim beamformer achieves “almost constant gains close to 0 dB.” (Ex. 1003, 90; Ex. 1002 ¶96.) A gain of zero means that subtracting the second virtual microphone’s noise-only output from the first virtual microphone’s target-plus-noise output will produce the target signal in the direction of the target, demonstrating the two virtual microphones have different responses to the target signal. (Ex. 1002 ¶96.)

In contrast, as highlighted in red, “in the direction of the interference, a deep null is formed.” (Ex. 1003, 90; Ex. 1002 ¶97.) The deep null reflects that subtracting the two virtual microphones’ outputs cancels the interference, which confirms that the virtual microphones’ responses to the interference are essentially the same. (Ex. 1002 ¶97.)

**e. A Signal Processor Operative to Combine Microphone Signals by Filtering and Summing in the Time Domain**

Claim element 1[c] recites “a signal processor coupled with the first and second microphone signals and operative to combine the first and second microphone signals by filtering and summing in the time domain.” Brandstein discloses this limitation.

Figure 3 of the '357 patent illustrates a virtual microphone formed from **filtering** and **summing** physical-microphone signals:

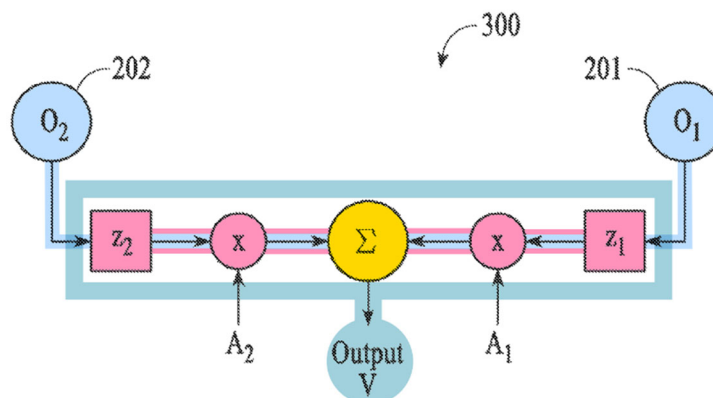
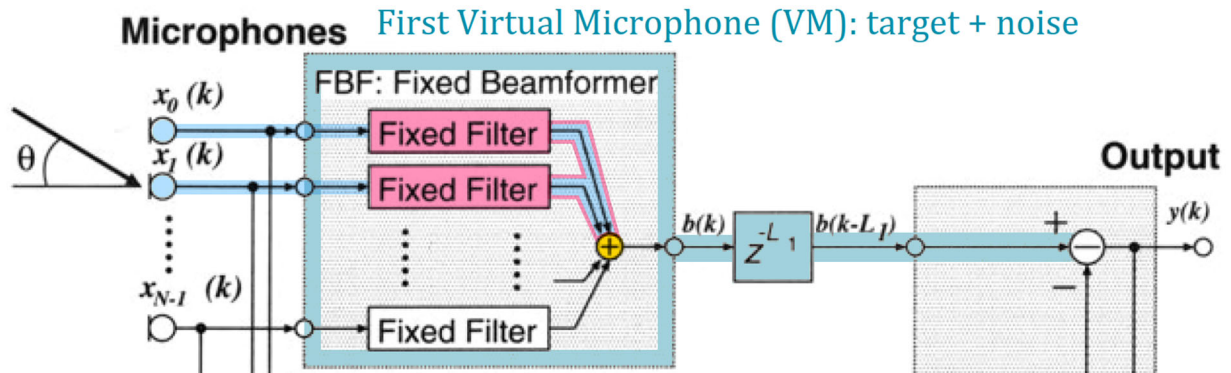


FIG.3

(Ex. 1001, Fig. 3.) Figure 4 illustrates the same concept with two processing paths to form two virtual microphones. (*Id.*, Fig. 4.) Consistent with Figures 3 and 4, claim 1 permits either virtual microphone or both to be formed by the recited filtering and summing. The '357 patent acknowledges this was a “common” technique known in the art for constructing virtual microphones. (*Id.*, 8:55-61.)

In Brandstein, the virtual microphones in the top and bottom branches of Figure 5.1 are both formed by a signal processor combining **two physical-microphone signals** by **filtering** and **summing**. (Ex. 1002 ¶100.) For example, below is the top branch:



(Ex. 1003, 89, Fig. 5.1 (excerpt).) As shown, the fixed beamformer combines the microphone signals in the top branch by applying fixed filters and summing. (Ex. 1002 ¶100.) Because Figure 5.1 shows the microphone signals as functions of discrete time  $k$ , it shows the filtering and summing as occurring in the time domain. (*Id.*)

Even if Brandstein’s Figure 5.1 had not disclosed filtering and summing *in the time domain*, it would have been obvious to a POSITA to implement the GSC’s filtering and summing in the time domain because performing those operations in the time domain would simply be a design choice between two well-known options: the time domain or a transform domain (also called “frequency domain”). (*Id.* ¶101.) Microphones output signals in the time domain, and the time domain has historically

been the default domain in which to implement filters, so filtering and summing in the time domain would be a natural choice. (*Id.*) Indeed, Lindemann notes that using the GSC in a microphone array for hearing aids was a “time domain approach.” (Ex. 1007, 1:51-52.)

Finally, it would have been obvious, if not inherent, that this processing of the microphone signals would be performed by a signal processor coupled with the signals. (Ex. 1002 ¶102.) Brandstein’s title is *Microphone Arrays: Signal Processing Techniques and Applications*. And Brandstein discloses implementing a Griffiths-Jim beamformer on a system comprising “a microphone array and a compact touch-panel personal computer which includes a floating point DSP, the ADSP-21062.” (Ex. 1003, 104 (internal citation omitted); *see also id.*, 105, Fig. 5.16.) Here, “DSP” is an abbreviation for digital signal processor. (Ex. 1002 ¶102.) A POSITA would have understood that some signal processor, such as the ADSP-21062, would perform the operations on the microphone signals shown in Figure 5.1 and that the processor was thus coupled to the microphone signals. (*Id.*)

**f. Applying a Varying Linear Transfer Function**

Claim element 1[c] then recites the signal processor is operative to “apply a varying linear transfer function between the first and second microphone signals.”

Gannot discloses applying a varying linear transfer function between the first and second microphone signals. Gannot sets forth a more-robust GSC solution that

can handle source-to-microphone acoustic paths with arbitrary transfer functions.

Gannot's Figure 3 shows the proposed GSC structure, and Figure 4 summarizes the algorithm:

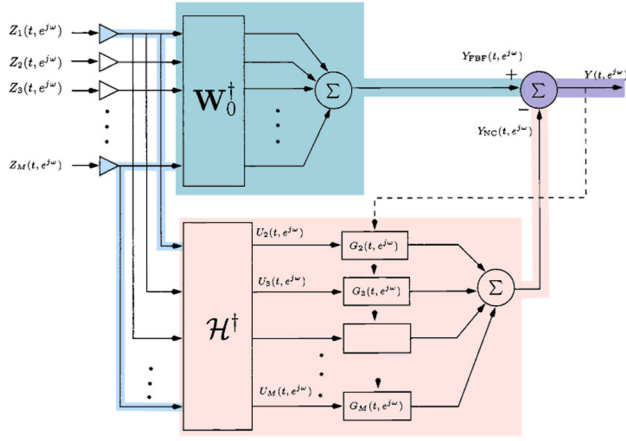


Fig. 3. Linearly constrained adaptive beamformer.

- 1) TF-s ratios:  $\mathbf{H}(e^{j\omega}) = \frac{\mathbf{A}(e^{j\omega})}{A_1(e^{j\omega})}$
  - 2) Fixed beamformer:  
 $Y_{\text{FBF}}(t, e^{j\omega}) = \mathbf{W}_0^\dagger(e^{j\omega})\mathbf{Z}(t, e^{j\omega})$
  - 3) Noise reference signals:  
 $\mathbf{U}(t, e^{j\omega}) = \mathcal{H}^\dagger(e^{j\omega})\mathbf{Z}(t, e^{j\omega})$
  - 4) Output signal:  
 $Y(t, e^{j\omega}) = Y_{\text{FBF}}(t, e^{j\omega}) - \mathbf{G}^\dagger(t, e^{j\omega})\mathbf{U}(t, e^{j\omega})$
  - 5) Filters update, for  $m = 1, \dots, M-1$ :  
 $\tilde{G}_m(t+1, e^{j\omega}) = G_m(t, e^{j\omega}) + \mu \frac{U_m(t, e^{j\omega})Y^*(t, e^{j\omega})}{P_{\text{est}}(t, e^{j\omega})}$   
 $G_m(t+1, e^{j\omega}) \xrightarrow{\text{FIR}} \tilde{G}_m(t+1, e^{j\omega})$
- where,  
 $P_{\text{est}}(t, e^{j\omega}) = \rho P_{\text{est}}(t-1, e^{j\omega}) + (1-\rho) \sum_m |Z_m(t, e^{j\omega})|^2$   
6) keep only non-aliased samples.  
(note:  $\mathbf{W}_0(e^{j\omega})$  is defined in (16).  
 $\mathcal{H}(e^{j\omega})$  is defined in (19)).

Fig. 4. Suggested algorithm.

Ex. 1004, Fig. 3

Ex. 1004, Fig. 4

The transfer functions Gannot addresses are linear and time-varying. (Ex. 1002 ¶105; Ex. 1004, 1615-1617 (“time-varying TFs” and represented by linear systems).) To account for the arbitrary acoustic paths, Gannot incorporates the corresponding transfer functions into the GSC algorithm and applies them. (Ex. 1002 ¶105.) For example, the blocking matrix  $\mathcal{H}^\dagger$  is used to create noise reference signals  $U_m$  that apply the linear transfer functions  $A_m$  of the acoustical paths:

$$U_m(e^{j\omega}) = Z_m(t, e^{j\omega}) - \frac{A_m(e^{j\omega})}{A_1(e^{j\omega})} Z_1(t, e^{j\omega})$$

$$m = 2, \dots, M. \quad (20)$$

(Ex. 1004, 1618-20; Ex. 1002 ¶105.) By incorporating these terms, Gannot’s more-general GSC accounts for arbitrary transfer functions. In particular, by using the ratio of the transfer function for microphone *m* to the transfer function for microphone 1, Gannot’s algorithm applies a varying linear transfer function for an acoustical path between the two microphones. (Ex. 1002 ¶105.)

Referring to the original Griffiths-Jim GSC paper, Gannot starts with the original GSC in a microphone-array setting for speech applications, which is precisely what Brandstein discloses. (Ex. 1002 ¶106.) Gannot then proposes a modification to apply a varying linear transfer function between microphone signals to arrive at a more-robust GSC. (*Id.*) Gannot does not merely propose the concept of this more-robust GSC solution—Gannot sets forth the complete underlying algorithm and mathematics for the solution and also reports findings of a real-world implementation. (*Id.*; *see also generally* Ex. 1004.) Gannot thus not only expressly motivates modifying a standard microphone-array GSC as shown in Brandstein but also demonstrates that the modification succeeds. (Ex. 1002 ¶106.) A POSITA would have therefore been motivated to combine the teachings of Gannot and Brandstein and would have had a reasonable expectation of success. (*Id.*)

**g. Generating an Output Signal with Reduced Noise**

Claim element 1[c] further recites that the signal processor is operative to “generate an output signal having noise content that is attenuated with respect to speech content.” Brandstein discloses this limitation.

The overall **output** of Brandstein’s GSC is the target-plus-noise output of the first virtual microphone minus the noise-only output of the second virtual microphone. (Ex. 1002 ¶108.)

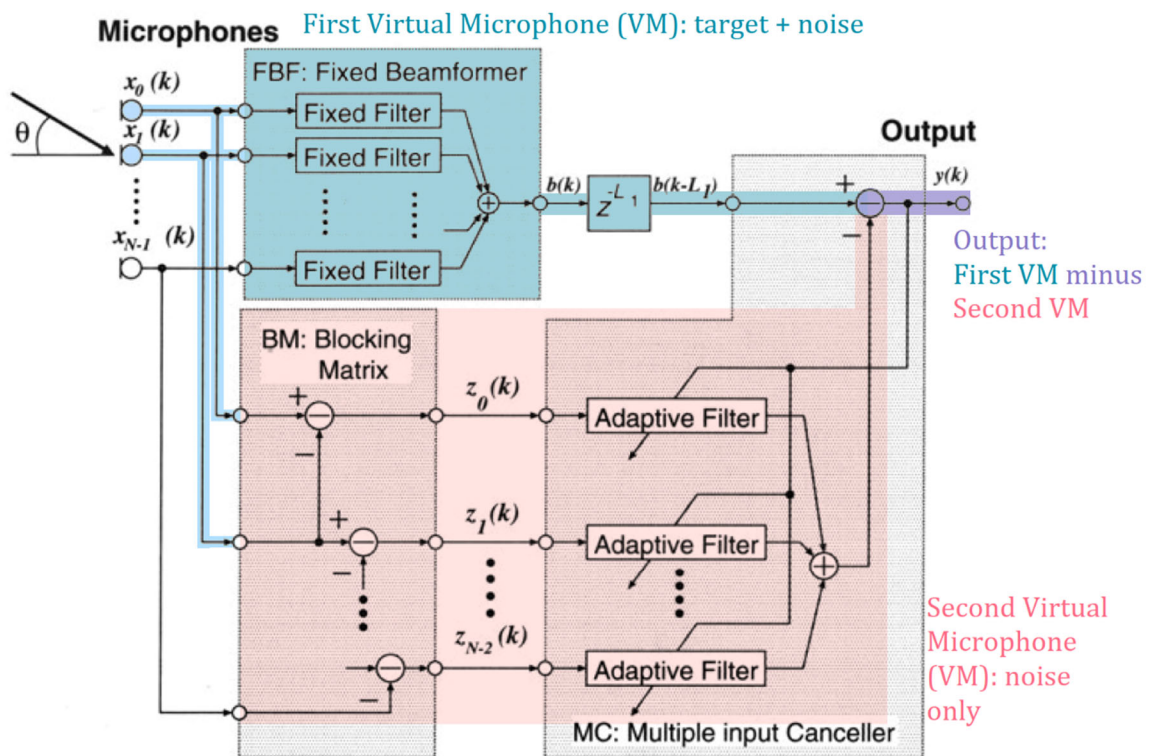


**Fig. 5.1.** Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceller (MC), and a blocking matrix (BM).

(Ex. 1003, 89, Fig. 5.1.)

The result is that, “in the subtracter output  $y(k)$ , the target signal is enhanced and undesirable signals such as ambient noise and interferences are suppressed.” (Ex. 1003, 88-89; Ex. 1002 ¶109.)

\* \* \*

In summary, both Brandstein and Gannot disclose a microphone-array GSC for reducing noise and enhancing speech signals. (Ex. 1002 ¶110.) Brandstein discloses the original Griffiths-Jim GSC in this microphone-array context, while Gannot discloses a more-robust GSC designed to handle source-to-microphone acoustical paths with varying linear transfer functions. (*Id.*) Gannot expressly teaches successfully modifying the original Griffiths-Jim GSC to apply ratios of transfer functions, with the ratios representing transfer functions of acoustical paths between microphones. (*Id.*) Brandstein discloses all other limitations of claim 1. (*Id.*) Because Gannot not only expressly proposes modifying the original GSC but explains exactly how to modify the original GSC and reports the successful results of a real-world implementation, claim 1 would have been obvious. (*Id.*)

## **2. Claim 2**

Claim 2 depends from claim 1 and further recites “wherein the signal processor comprises one or more digital signal processors (DSPs).” Brandstein discloses using a DSP to implement a GSC, and it would have been obvious to use a DSP in

any event. (Ex. 1003, 104, 105, Fig. 5.16; *see also supra* §VI.A.1.e; Ex. 1002 ¶111.)

Thus, claim 2 would have been obvious.

### 3. Claim 3

Claim 3 depends from claim 1 and further recites “wherein the noise content comprises acoustic noise and the speech content comprises human speech.” Brandstein discloses this limitation.

Brandstein teaches that noise can be acoustic noise like “airconditioner [sic] noise.” (Ex. 1003, 101; Ex. 1002 ¶113.) Brandstein discloses that, simultaneously, the target speech signal can be human speech, like “an English male speech signal.” (*E.g.*, Ex. 1003, 101.) Figure 5.13 illustrates this, with the “White Noise” source illustrating noise such as interference from an air conditioner:

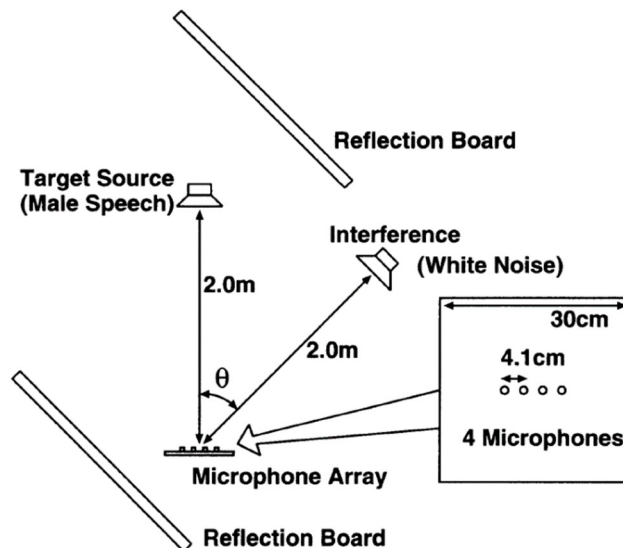


**Fig. 5.13.** Experimental set-up.

(*Id.*, 103, Figs. 5.13, 5.14; Ex. 1002 ¶113.)

Claim 3 would have been obvious.

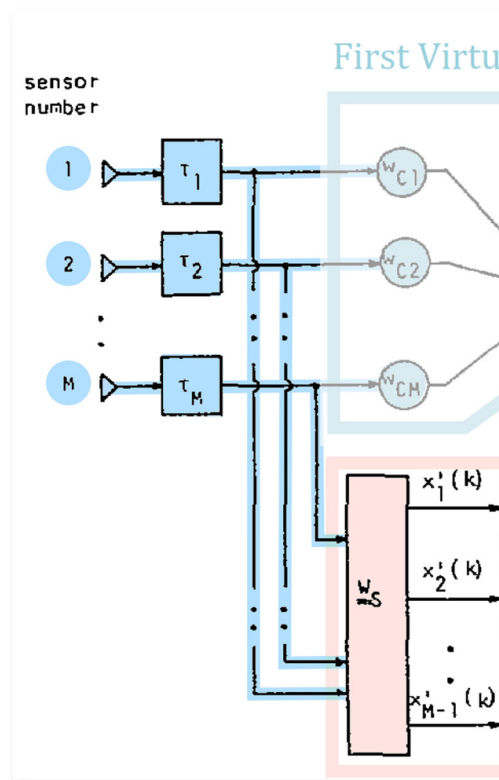
#### **4. Claim 4**

Claim 4 depends from claim 1 and further recites “wherein the signal processor is operative to add a delay to the first microphone signals.” Brandstein discloses this limitation.

Brandstein explains that the “BM [i.e., blocking matrix] in Fig. 5.1 has a simple delay-and-sum structure” (Ex. 1003, 90), which in the case of the blocking matrix is also called a “delay-and-subtract” structure since the signals are subtracted (summed with a negative sign) to remove the target signal (*id.*, 88-89). (Ex. 1002 ¶116.) In the case illustrated in Figure 5.1, the target signal’s assumed direction of arrival (the “look direction”) is perpendicular to the array surface, so “no delay element is necessary.” (Ex. 1003, 88-89; Ex. 1002 ¶116.) Thus, when  $\theta$  in Figure 5.1 is zero, the blocking matrix’s delay-and-subtract structure can simplify to “a set of subtractors ....” (Ex. 1003, 89; Ex. 1002 ¶116.) “This structure is actually the one shown in Fig. 5.1.” (Ex. 1003, 89.) But as Brandstein implies, a POSITA would have understood that, when the target signal is expected to arrive from a non-perpendicular direction (i.e., when  $\theta$  in Figure 5.1 is nonzero, as Figure 5.1 itself indicates is possible), the delay-and-subtract structure is not simplified. (Ex. 1002 ¶116.) Instead, before subtracting, delays are used to time-align the target-signal components of the microphone signals. (Ex. 1002 ¶116; *see also* Ex. 1003, 159 (“Delay-

and-sum beamformers apply time shifts to the array signals to compensate for the propagation delays in the arrival of the source signal at each microphone. These signals are time-aligned and summed together to form a single output signal.”.) This time alignment is what allows the subtraction operations to block the target signal. (Ex. 1002 ¶116.)

The time alignment implicit in Brandstein’s reference to a “delay-and-subtract” structure for the blocking matrix appears expressly in the original Griffiths-Jim paper:



(Ex. 1005, 30, Fig. 4 (excerpt); Ex. 1002 ¶117.) The elements labeled  $\tau_1, \dots, \tau_M$  are delays used to time-align the sensor signals. (See Ex. 1005, 29; Ex. 1002 ¶117.)

Griffiths-Jim thus expressly shows aligning the sensor signals before feeding them to the blocking matrix, just as Brandstein contemplates. (Ex. 1002 ¶117.) Indeed, Brandstein presents Figure 5.1 and the related discussion precisely to explain the original Griffiths-Jim paper. (*See id.*; Ex. 1003, 88-90, 108.) A POSITA reading Brandstein would have thus immediately understood that Brandstein’s reference to a “delay-and-subtract” structure for the blocking matrix means the signals should be aligned in time, which can be addressed through conventional delays of the kind Griffiths-Jim describes. (Ex. 1002 ¶117.)

Brandstein also expressly teaches modifying the original GSC to include additional delays to the microphone signals to compensate for errors in the assumed time alignment. (*See, e.g.*, Ex. 1003, 90-91 (motivating modification), 92-93 & Fig. 5.4 (introducing  $z^{-L_2}$  elements to delay microphone signals by  $L_2$  samples); Ex. 1002 ¶118.)

Claim 4 would have been obvious.

## **5. Claim 5**

Claim 5 depends from claim 4 and further recites “wherein the signal processor is operative to raise the delay 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.”

**a. Claim 5 Encompasses Standard Near-Field Time-Alignment.**

The '357 patent explains that “ $\gamma$  is the time difference between arrival of speech at  $V_1$  compared to  $V_2$ ” (Ex. 1001, 15:20-21), where 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  $\gamma$  is their difference divided by  $c$ , the speed of sound, and multiplied by the sampling frequency  $f_s$ ” (*id.*, 11:19-22). This appears in equation form in the '357 patent (below right), though there are a few typos (e.g.,  $f_x$  instead of  $f_s$ ); a provisional application (below left) incorporated by reference has the same equation without typos:

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

*Ex. 1011, 18*

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

*Ex. 1001, 11:11 (with typos)*

In other words, the '357 patent states the time difference between arrival of speech at  $V_1$  compared to  $V_2$  (i.e., the time difference claim 5 recites) is just the time difference between arrival of speech at  $O_1$  compared to  $O_2$ . (Ex. 1002 ¶121.)

This difference in arrival times is taken from the standard near-field equations and is denoted  $\tau_i$  in §I.E.2. (*Supra* §II.A.3 (comparing  $\gamma$  in '357 patent to  $\tau_i$  in §I.E.2); Ex. 1002 ¶121.) Standard near-field time-alignment can be accomplished

by applying a complex exponential term of the form  $e^{-j\omega\tau_i}$  to a microphone signal, which delays the signal in proportion to  $\tau_i$ . (Ex. 1002 ¶¶66-70, 121, 123, 126.)

Therefore, claim 5 encompasses delaying a signal in accordance with standard near-field time-alignment. (*Id.* ¶121; *also id.* ¶¶122-26.)

**b. Brandstein Discloses or Renders Obvious  
Standard Near-Field Time-Alignments for the  
GSC.**

Brandstein discloses the additional limitation in claim 5. Alternatively, at a minimum, the limitation would have been obvious from Brandstein's teachings.

As noted for claim 4, Brandstein teaches that the GSC's blocking matrix has a delay-and-sum (or delay-and-subtract) structure because the microphone signals are delayed so that their target-signal components are aligned in time, which is what allows the subtraction operations in the blocking matrix to block the target signal. (*Supra* §VI.A.4.) When the Griffiths-Jim paper published in 1982, delay-and-sum beamforming was already well developed in the art, and POSITAs understood why Griffiths-Jim disclosed delay elements to time-align signals before feeding them to the blocking matrix. (Ex. 1002 ¶122.) By 2007, skilled artisans had known for decades the role time-alignment plays in the lower branch of the GSC. (*Id.*)

At the same time, POSITAs would have been long aware of far-field versus near-field distinctions and how to time-align signals from near-field sources. (*Id.* ¶¶62-70, 123.) In fact, applying near-field time-alignment for the lower branch of

the GSC would have been part of a POSITA's background knowledge because McCowan disclosed exactly that in 2002:

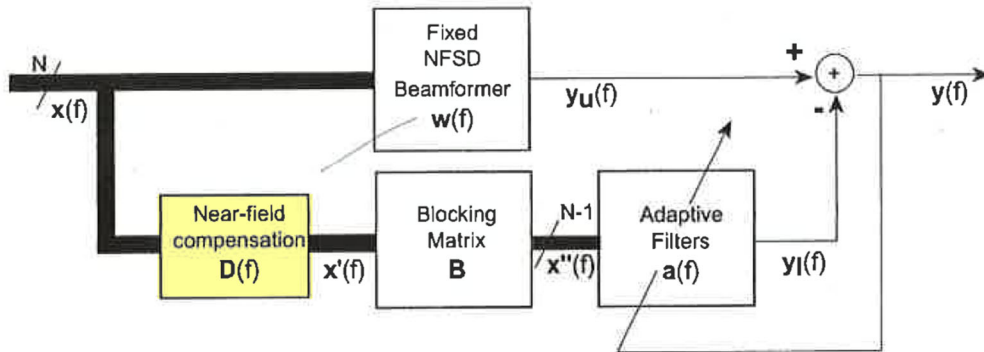


FIG. 3. Near-field adaptive beamformer.

(Ex. 1006, 91, Fig. 3; Ex. 1002 ¶123.) As McCowan discloses, to ensure that the standard GSC's blocking matrix can fulfill its purpose of completely blocking the desired signal, “the desired signal must ... be time aligned[.]” (Ex. 1006, 93.) For near-field sources, McCowan teaches introducing a near-field compensation unit that time-aligns the microphone signals by applying to each microphone output a complex exponential of the form taught in Brandstein's equation 2.28 (discussed in §I.E.2 and also below). (*Id.*, 93 (discussing equation 16 and referring to equation 6); Ex. 1002 ¶¶68-70, 123.) This complex exponential satisfies the limitation claim 5 adds to claim 4. (Ex. 1002 ¶¶121, 123, 126.) “Once this near-field compensation has been performed, a standard GSC blocking matrix can be employed to block the desired signal from the adaptive path.” (Ex. 1006, 93; *see also id.*, 95

(“The blocking matrix and adaptive noise canceling filters are taken from the standard GSC technique [citation]” (citing Griffiths-Jim)).)

In view of this background knowledge, claim 5 would have been obvious to a POSITA based on Brandstein’s Chapter 5 GSC discussion alone. A POSITA would have known that some applications involve near-field target signals, that the GSC’s blocking matrix requires time-alignment, that near-field time-alignment involves applying delays encompassed by claim 5, and that McCowan discloses such near-field time-alignment followed by a standard GSC blocking matrix. (Ex. 1002 ¶124.) The motivation and knowledge to use near-field time-alignment to implement the delays for the delay-and-subtract blocking matrix in the original Griffiths-Jim GSC (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).

Alternatively, to the extent PO argues that such near-field time-alignment would require modifying the GSC described in Chapter 5, the modification represents nothing more than the simple addition of one known element (near-field time alignment) to another known element (the GSC) 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). (Ex. 1002 ¶124.) The combination also applies a known technique (time-alignment for near-field

sources) to a well-known device (GSC) that is ready for improvement, with the combination yielding predictable results. (*Id.*) A POSITA would have had a reasonable expectation of success because, as McCowan demonstrates, near-field time-alignment with the GSC was trivial and shown to work. Claim 5 would therefore have been obvious in view of Brandstein’s Chapter 5 and the background knowledge of a POSITA. *KSR*, 550 U.S. at 416-17.

Alternatively, claim 5 would have been obvious in view of the combination of Brandstein’s discussion of the GSC and near-field time-alignment in Chapters 2 and 5. In Chapter 5, Brandstein explains that the GSC blocking matrix has a delay-and-subtract (delay-and-sum with negative coefficients) structure: “The BM in Fig. 5.1 has a simple delay-and-sum structure” (Ex. 1003, 90); and the “simplest structure for the BM is a delay-and-subtract beamformer which was described in the previous section” (Ex. 1003, 88). It is evident the “previous section” referred to is Chapter 2, which is the last chapter of Brandstein before Chapter 5 to discuss delay-and-sum/subtract structures and the GSC blocking matrix. (Ex. 1002 ¶125.)

In Chapter 2, specifically section 2.3, Brandstein discloses that time-aligning signals for near-field beamforming takes into account the 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.

(Ex. 1003, 25; Ex. 1002 ¶126.) As noted previously (*supra* §§I.E.2, II.A.3),  $\tau_i$  is the time difference in arrival between microphone  $i$  and the reference microphone. (Ex. 1002 ¶126.) Put another way,  $\gamma$  in the '357 patent is just  $\tau_i$  in Brandstein's notation for the standard near-field design (and just expressed in different units of time). (*Id.*)

Assuming near-field conditions, and using Brandstein's notation, 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.*) To time-align the  $x_i$  signals, the inverse delay is applied: applying the complex-exponential term specified in Brandstein equation 2.28,  $e^{-j\omega\tau_i}$ , to  $x_i$  (i.e.,  $e^{-j\omega\tau_i}x_i$ ), delays  $x_i$  in proportion to  $\tau_i$  to time-align the signals. (*Id.*)

In section 2.4, Brandstein discloses “all designs presented in section 2.3 can be implemented by using the GSC-structure.” (Ex. 1003, 35.) Specifically, a “multi-channel *time-aligned* input signal ... is projected onto the noise-only subspace<sup>[footnote]</sup> by a blocking matrix  $\mathbf{B}$ ,” with the footnote stating: “Which means that the desired signal is spatially filtered out (blocked).” (*Id.*, 34 & n.5 (emphasis added).) A “well-known example” of  $\mathbf{B}$  “is the original Griffith[s]-Jim matrix which subtracts two adjacent channels.” (*Id.*) In summarizing Chapter 2, section 2.5 states: “we emphasized the close relationship between superdirective arrays and adaptive beamformers and their well-known implementation as a generalized sidelobe canceler,” as well as that “nearfield assumptions ... can be easily included.” (*Id.*, 36-37.)

Chapter 2 therefore provides express motivation to use Brandstein’s disclosure of near-field time-alignment in section 2.3 to implement the delay for the GSC’s blocking matrix. Furthermore, in Chapter 13, Brandstein explains that delay-and-sum beamforming was “first developed on the basis of the far-field assumption, but may also be extended to near-field beamforming.” (Ex. 1003, 290 (internal citation omitted).) A POSITA would have understood that Brandstein teaches applying the complex-exponential term specified in Brandstein equation 2.28 to delay a signal in proportion to  $\tau_i$  to time-align the signals by disclosing: (1) the blocking matrix has a delay-and-sum (or delay-and-subtract) structure; (2) the purpose of the delay in delay-and-sum beamforming is to time-align the sensor signals; (3) delay-and-sum

beamforming can be extended to near-field designs; and (4) the standard near-field design specifies a delay whose magnitude is equal to the difference in propagation time of the target signal to one microphone relative to another (reference) microphone. (Ex. 1002 ¶127.) Combining the teachings of Chapters 2 and 5 to arrive at claim 5 is a simple addition of one known element (near-field time alignment) to another known element (the GSC) to obtain predictable results (effective target-signal blocking in the lower branch). The combination also applies a known technique of time-alignment for near-field sources to the well-known GSC that is ready for improvement, with the combination yielding predictable results. (*Id.*) Again, a POSITA would have had a reasonable expectation of success, particularly in light of McCowan, and claim 5 would therefore have been obvious in view of Brandstein’s Chapters 2 and 5 and the background knowledge of a POSITA. *KSR*, 550 U.S. at 416-17.

## **6. Claim 6**

Claim 6 depends from claim 4 and further recites “wherein the signal processor is operative to raise the delay to a power that is proportional to a sampling frequency multiplied by a quantity equal to a third distance subtracted from a fourth distance, the third distance being between the first physical microphone and a speech source of the speech and the fourth distance being between the second physical mi-

crophone and the speech source.” Brandstein discloses this limitation. Alternatively, at the very least, it would have been obvious in view of Brandstein’s teachings.

As discussed above for claim 5, Brandstein renders obvious extending the standard GSC to a near-field design, including by time-aligning the microphone signals fed to the blocking matrix by applying delay term  $e^{-j\omega\tau_i}$ , whose exponent is proportional to  $\tau_i$ . (*Supra* §VI.A.5; Ex. 1002 ¶129.) Brandstein defines  $\tau_i$  as:

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

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.

(Ex. 1003, 25.) Brandstein’s  $\tau_i$  is thus “a quantity equal to a third distance subtracted from a fourth distance, the third distance being between the first physical microphone and a speech source of the speech and the fourth distance being between the second physical microphone and the speech source,” divided by the speed of sound. (Ex. 1002 ¶129.) Claim 6 recites an exponent “proportional” to “a sampling frequency multiplied by” a quantity such as Brandstein’s  $\tau_i$ . (*Id.*) But  $\tau_i$  is proportional to the sampling frequency multiplied by  $\tau_i$ , so Brandstein’s delay term  $e^{-j\omega\tau_i}$  satisfies claim 6. (*Id.*)

Alternatively, while Brandstein discloses the delay term in the form above with time expressed in seconds, it would have also been routine for a POSITA to rewrite the formula to express time in terms of number of samples, as is common for digital signal processing, which involves discrete sampling. (*Id.* ¶130.) To convert from seconds to samples, one simply multiplies the delay expressed in seconds by the sampling frequency  $f_s$  expressed in samples per second. (*Id.*) For example, after that trivial operation, Brandstein’s delay term  $e^{-j\omega\tau_i}$  becomes  $e^{-j\omega\tau_i f_s}$ . (*Id.* (explaining that, using z-transform notation, this is also commonly written as  $z^{-\tau_i f_s}$ .) Applying that exponential term delays a signal by  $\tau_i f_s$  samples, as desired. (*Id.*)

A POSITA would have been motivated to rewrite Brandstein’s near-field delay in terms of samples. Brandstein itself motivates rewriting this delay element in terms of samples for use with the GSC, as Figure 5.1 is shown in terms of samples. (*Id.* ¶131; Ex. 1003, 88-89 & n.1 (discussing effect of  $z^{-L_1}$  block in figure as an “ $L_1$ -sample delay”).) Brandstein also discloses implementing the GSC with a digital signal processor. (Ex. 1003, 104 & Fig. 5.16.) Additionally, digital signal processing was a known technique, and a POSITA would have been motivated to implement the GSC digitally because, among other advantages over analog systems, digital circuits can be reproduced more easily in large quantities at low cost, digital systems are not affected by changes of component values in analog systems caused by factors such as age, temperature, and humidity, digital signal processing allows

for flexibility through digitally programmable systems, digital systems are generally more accurate, digital signals can be easily stored on media using semiconductor chips, and digital signals can be transmitted over long distances. (Ex. 1002 ¶131.) Converting Brandstein’s delay formula from seconds to samples for digital signal processing would have involved the simple and well-known operation of multiplying by the sampling frequency to convert time units from, e.g., seconds to samples. (*Id.*)

Claim 6 would have therefore been obvious.

#### **7. Claim 7**

Claim 7 depends from claim 1 and further recites “wherein the first and second physical microphones comprise omnidirectional microphones.” Brandstein discloses this limitation. (Ex. 1003, 101 (implementation using “omni-directional microphones”); Ex. 1002 ¶¶89, 133.) Claim 7 would have been obvious.

#### **8. Claim 8**

Claim 8 depends from claim 1 and further recites “wherein the first and second physical microphones are included in a microphone array.” Brandstein discloses this limitation. The first and second microphones in Brandstein’s Figure 5.1 are included in a microphone array. (Ex. 1003, 88 (referring to “array surface”), 90 (“With the same microphone array ....”); *also id.*, 103, Fig. 5.13; Ex. 1002 ¶134.) Brandstein is a reference dedicated to microphone arrays, and the GSC is a technique

used with arrays. (Ex. 1002 ¶134; Ex. 1003, 88-90 (referring to “array[s]” in discussing GSC).) Claim 8 would have been obvious.

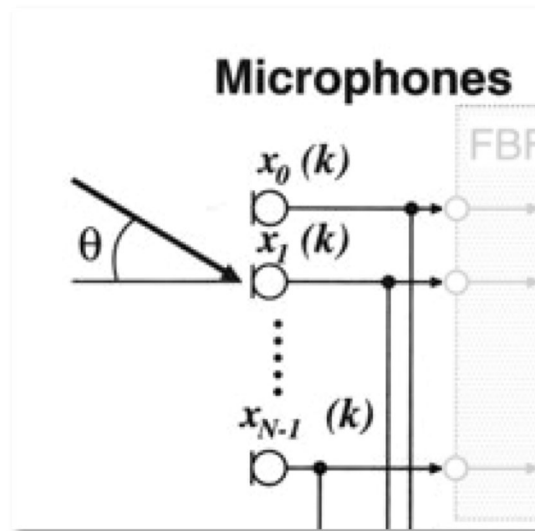
### **9. Claim 9**

Claim 9 depends from claim 1 and further recites “wherein the first physical microphone and the second physical microphones are disposed along an axis and are separated from each other by a first distance.” Brandstein discloses this limitation. For example, the microphones used in Figures 5.1, 5.13, and 5.16 are shown as lying on the same axis, separated by some distance. (Ex. 1003, 89 (Fig. 5.1), 103 (microphones in Fig. 5.13 shown in linear array with 4.1 cm spacing), 105 (indicating “Upper Row” of microphones used in Fig. 5.16); Ex. 1002 ¶136.) In general, most array techniques require or assume the microphones are spaced apart, and it is inherent that a line can be drawn between any two spatially separated microphones. (Ex. 1002 ¶136.) Thus, claim 9 would have been obvious.

### **10. Claim 10**

Claim 10 depends from claim 9 and further recites “wherein a midpoint of the axis is a second distance from a speech source that generates the speech, wherein the speech source is located in a direction defined by an angle relative to the midpoint.” Claim 10 adds nothing meaningful to claim 9. Claim 9 assumes the microphones are disposed along an axis and are separated by a first distance. Any distance has a midpoint. Claim 1 discusses speech, which must originate from a source. And any

speech source is inherently some distance from the midpoint and at some angle relative to the midpoint. Claim 10 would have therefore been obvious for the same reasons claim 9 would have been. (Ex. 1002 ¶138.) In any event, the additional limitations of claim 10 would have been obvious in view of Brandstein because Figure 5.1 discloses a linear microphone array along an axis and the target speech signal arriving at an angle  $\theta$ :



(Ex. 1003, 89, Fig. 5.1 (excerpt); Ex. 1002 ¶139.) Under the far-field assumption, the speech signal's wavefront is a plane that will arrive at the same angle at all points along the array axis. (Ex. 1002 ¶139; Ex. 1003, 3.)

Claim 10 would have been obvious.

## **11. Claim 11**

Claim 11 depends from claim 10 and further recites “wherein the first virtual microphone is formed by subtracting the second microphone signal from the first microphone signal.” Brandstein discloses this limitation.

In claim 1, it does not matter which virtual microphone, or branch, in the GSC is labeled the “first” versus the “second.” For claim 1, Petitioner referred to the top branch as the “first” virtual microphone and the bottom branch as the “second” virtual microphone.

With a mere switch of the labels, the same disclosure renders claim 11 obvious. For claim 11, the top branch of the GSC is the recited “second” virtual microphone and the bottom branch is the recited “first” virtual microphone. Then the **first virtual microphone** (now the lower branch) includes the blocking matrix with its delay-and-subtract structure that **subtracts** the **second microphone signal** from the **first microphone signal** as recited in claim 11:

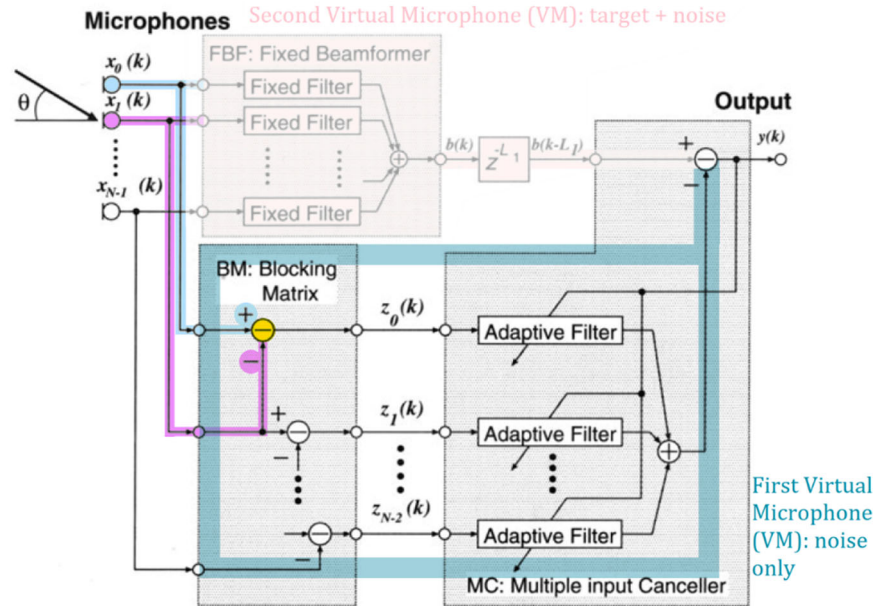


**Fig. 5.1.** Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceller (MC), and a blocking matrix (BM).

(Ex. 1003, 89, Fig. 5.1; Ex. 1002 ¶143.)

Claim 11 would have therefore been obvious.

## 12. Claim 12

Claim 12 depends from claim 10 and further recites “wherein the second virtual microphone is formed by subtracting the first microphone signal from the second microphone signal.” Brandstein discloses the additional limitation recited in claim 12. Consistent with Petitioner’s discussion of claim 1 (and claims 9 and 10), the “second virtual microphone” is the bottom branch of the GSC.

For claim 12, microphone signal  $x_1$  is the “first” microphone signal and microphone signal  $x_0$  is the “second.” (Ex. 1002 ¶146.) With this designation, the **second virtual microphone** includes the blocking matrix with its delay-and-subtract

structure that **subtracts** the **first microphone signal** from the **second microphone signal**:

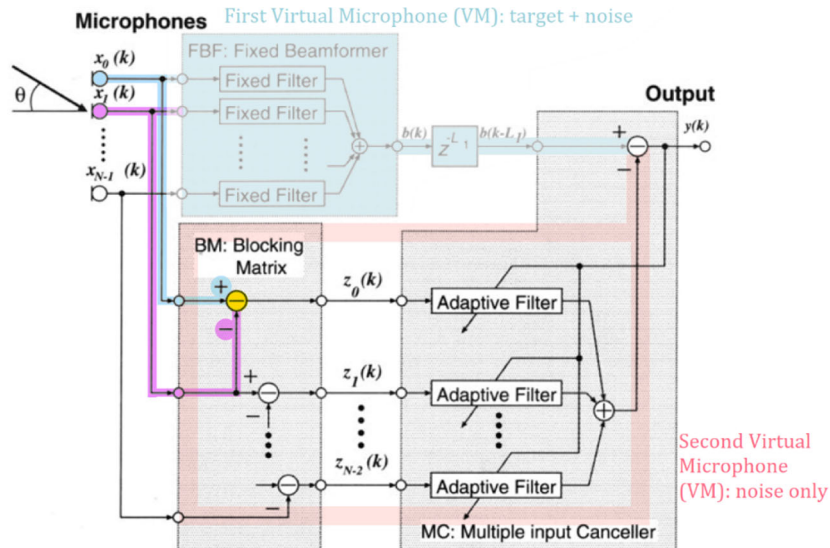


**Fig. 5.1.** Griffiths-Jim beamformer. It comprises a fixed beamformer (FBB), a multiple-input canceller (MC), and a blocking matrix (BM).

(Ex. 1003, 89, Fig. 5.1; Ex. 1002 ¶146.)

Claim 12 would have therefore been obvious.

### 13. Claim 13

Claim 13 depends from claim 1 and further recites “wherein the first virtual microphone is formed by subtracting the second microphone signal from a delayed version of the first microphone signal.” Brandstein discloses this limitation or at least renders it obvious. For this claim, as in claim 11, the bottom branch of the GSC is the “first virtual microphone.”

As discussed, Brandstein discloses or renders obvious that, if the target signal is expected to arrive from a direction not perpendicular to the array surface, the non-

reference microphone signals will be delayed to be time-aligned before performing the subtraction in the blocking matrix that blocks the target signal—hence, Brandstein’s statement that the blocking matrix has a “delay-and-subtract” structure. (*Supra* §§VI.A.4-VI.A.6; Ex. 1002 ¶150.) When the second microphone signal,  $x_1$ , is the reference signal, a delay is applied to the first microphone signal,  $x_0$ , to time-align it, and no delay is applied to the second microphone signal. (Ex. 1002 ¶150.)

Thus, Brandstein discloses or renders obvious a **first virtual microphone** (that includes the blocking matrix with its delay-and-subtract structure) that **subtracts** the **second microphone signal** from a **delayed version of the first microphone signal**.

(*Id.*)

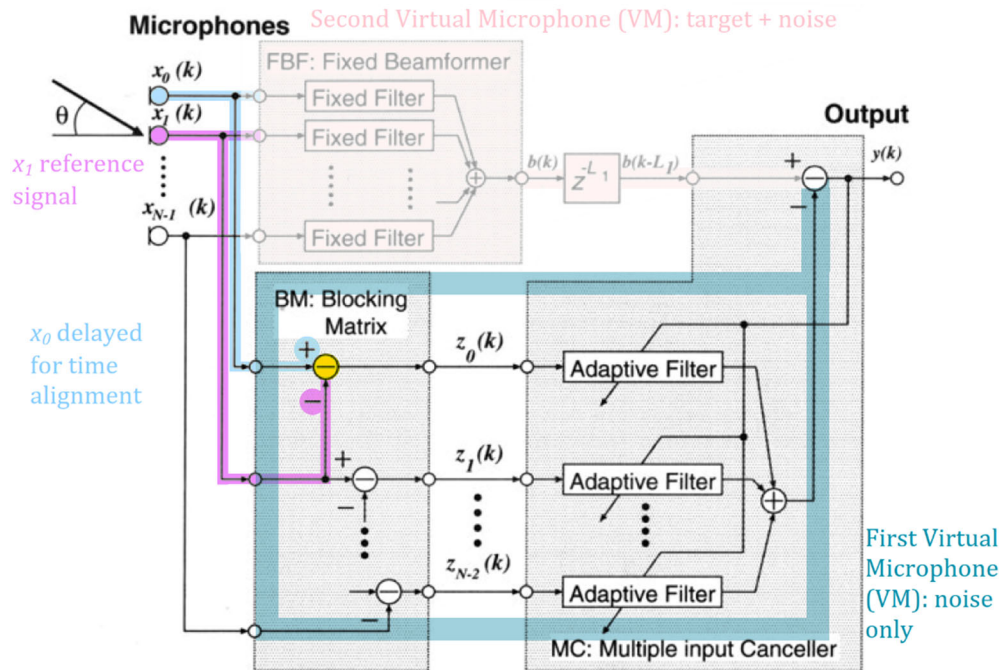


**Fig. 5.1.** Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceller (MC), and a blocking matrix (BM).

(Ex. 1003, 89, Fig. 5.1.)

Claim 13 would have been obvious.

#### 14. Claim 14

Claim 14 depends from claim 1 and further recites “wherein the second virtual microphone is formed by subtracting the first microphone signal from a delayed version of the second microphone signal.” Brandstein discloses or renders obvious this limitation. The analysis for claim 14 is the same as for claim 13, except that the

bottom branch of Brandstein’s GSC is the “second virtual microphone,” and the designation of the “first” and “second” microphone signals is reversed. Thus, the claim would have been obvious for the same reasons as claim 13. (Ex. 1002 ¶153.) Specifically, when the bottom branch of the GSC is the recited “second virtual microphone” and the “first” microphone signal ( $x_1$ ) is the reference signal, a delay is applied to the second microphone signal ( $x_0$ ) to time-align it, and no delay is applied to the first microphone signal. (*Id.* ¶154.) Thus, Brandstein discloses or renders obvious that the **second virtual microphone** (which includes the blocking matrix with its delay-and-subtract structure) **subtracts** the **first microphone signal** from a **delayed version of the second microphone signal** as shown:

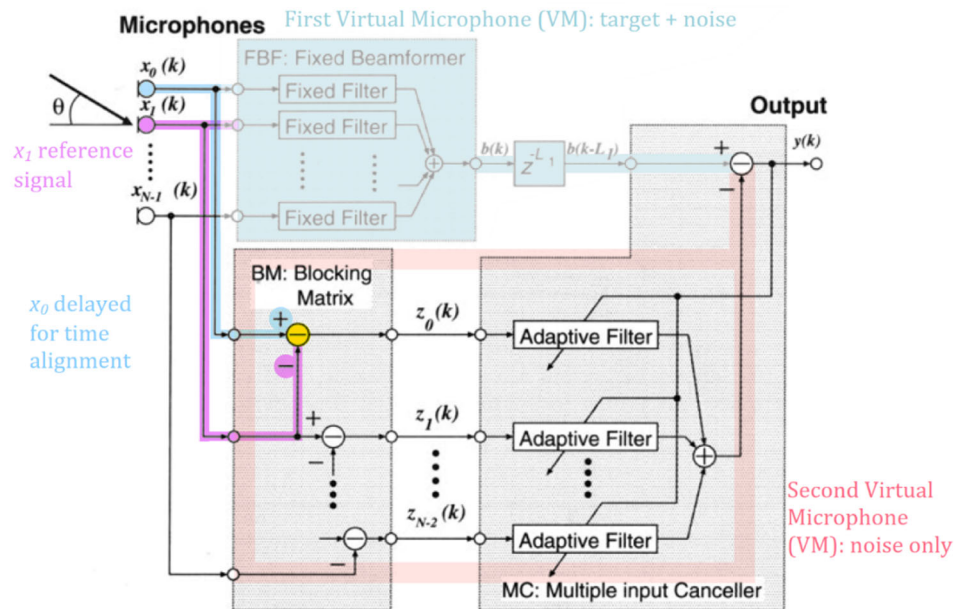


**Fig. 5.1.** Griffiths-Jim beamformer. It comprises a fixed beamformer (FBF), a multiple-input canceller (MC), and a blocking matrix (BM).

(Ex. 1003, 89, Fig. 5.1; Ex. 1002 ¶154.)

Claim 14 would have been obvious.

## **15. Claim 15**

### **a. Preamble**

The preamble recites “a device, comprising ....” Brandstein discloses this limitation. (*Supra* §VI.A.1.a; Ex. 1002 ¶¶85-86, 156.)

### **b. First Virtual Microphone**

Claim element 15[a] recites “a first virtual microphone comprising 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.” As discussed for claim 1, Brandstein discloses this limitation. (*Supra* §VI.A.1.b; Ex. 1002 ¶157.)

### **c. Second Virtual Microphone**

Claim element 15[b] recites “a second virtual microphone comprising a second combination of the first microphone signal and the second microphone signal, wherein the second combination is different from the first combination.” As discussed for claim 1, Brandstein discloses this limitation. (*Supra* §VI.A.1.c; Ex. 1002 ¶158.)

**d. Substantially Similar Responses to Noise and Substantially Dissimilar Responses to Speech**

Claim element 15[b] further recites “wherein the first virtual microphone and the second virtual microphone are distinct virtual directional microphones with substantially similar responses to noise and substantially dissimilar responses to speech.” As discussed for claim 1, Brandstein discloses this limitation. (*Supra* §VI.A.1.d; Ex. 1002 ¶159.)

**e. Virtual Microphone Array with a Single Null**

Claim element 15[c] recites “a virtual microphone array including the first and second virtual microphones and having a single null oriented in a direction toward a source of speech.” Brandstein discloses this limitation.

Brandstein discloses that the first and second virtual microphones (top and bottom branches of the GSC) output separate signals that, in accordance with common array processing, are combined with each other. (Ex. 1002 ¶161.) The output of the bottom branch is subtracted from the output of the top branch. (*Id.*) Only the bottom branch (second virtual microphone) has a null oriented in the direction of the target speech signal:

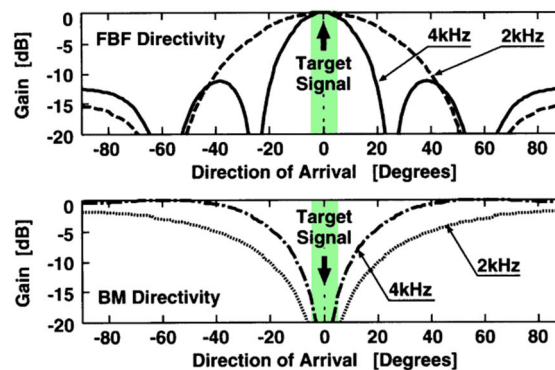


**Fig. 5.3.** Directivity pattern of a fixed beamformer (FBF) and a blocking matrix (BM).

(Ex. 1003, 90, Fig. 5.3; Ex. 1002 ¶161.) Whereas the second virtual microphone has a null in the direction of the target speech, the first virtual microphone (top branch) does not. (Ex. 1002 ¶161.) Brandstein thus discloses a virtual microphone array including the first and second virtual microphones and having a single null oriented in a direction toward a source of speech. (*Id.*)

#### **f. Signal Processor**

Claim element 15[d] begins by reciting “a signal processor coupled with the first and second microphone signals and operative to combine the first and second microphone signals by filtering and summing in the time domain.” As discussed for claim 1, Brandstein discloses or renders obvious this limitation. (*Supra* §VI.A.1.e; Ex. 1002 ¶162.)

#### **g. Applying a Varying Linear Transfer Function**

Claim element 15[d] further recites the signal processor is operative to “apply a varying linear transfer function between the first and second microphone signals.”

As discussed for claim 1, Gannot discloses this limitation, and a POSITA would have been motivated to combine Brandstein with Gannot with a reasonable expectation of success. Thus, this limitation would have been obvious in view of Brandstein and Gannot. (*Supra* §VI.A.1.f; Ex. 1002 ¶163.)

**h. Generating an Output Signal with Reduced Noise**

Claim element 15[d] further recites the signal processor is operative to “generate an output signal having noise content that is attenuated with respect to speech content.” As discussed for claim 1, Brandstein discloses this limitation. (*Supra* §VI.A.1.g; Ex. 1002 ¶164.)

\* \* \*

In summary, both Brandstein and Gannot disclose a microphone-array GSC for reducing noise and enhancing speech signals. (Ex. 1002 ¶165.) Brandstein discloses the original Griffiths-Jim GSC in this microphone-array context, while Gannot discloses a more robust GSC designed to handle source-to-microphone acoustical paths with arbitrary transfer functions. (*Id.*) Gannot expressly teaches modifying the original GSC to apply ratios of transfer functions, with the ratios representing transfer functions of acoustical paths between microphones. (*Id.*) Brandstein discloses all other limitations of claim 15. (*Id.*) Because Gannot not only expressly

proposes modifying the original GSC but explains exactly how to modify the original GSC and reports the successful results of a real-world implementation, claim 15 would have been obvious. (*Id.*)

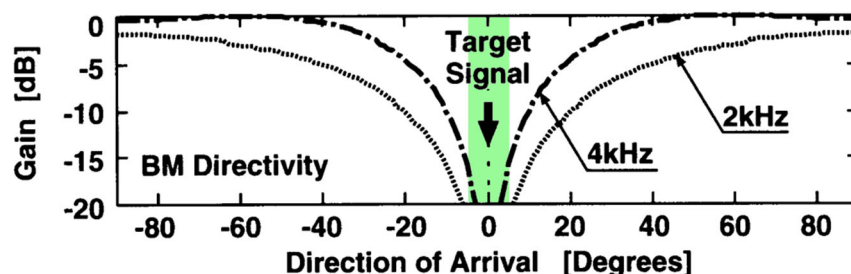
## 16. Claim 16

Claim 16 depends from claim 15 and further recites “wherein the source of speech comprises human speech.” As discussed for claim 3, which is similar, Brandstein discloses that the speech can be human speech. (*Supra* §VI.A.3; Ex. 1002 ¶166.) Claim 16 would have been obvious.

## 17. Claim 17

Claim 17 depends from claim 15 and further recites “wherein the second virtual microphone includes a second linear response to speech and the single null comprises a region of the second 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.” Brandstein discloses this limitation.

The second virtual microphone (the bottom branch of the GSC) has a null in the direction of the speech signal, as the second diagram in Figure 5.3 illustrates:



(Ex. 1003, 90, Fig. 5.3 (second diagram); Ex. 1002 ¶169.) The second virtual microphone’s response to the speech is linear because it can be represented by a linear system. (Ex. 1002 ¶169.) Figure 5.3 shows the null is a true minimum for the second virtual microphone. (*Id.*) Therefore, the null comprises 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 linear response to speech. (*Id.*)

Claim 17 would have been obvious.

#### **18. Claim 18**

Claim 18 depends from claim 15 and further recites “wherein the first virtual microphone is formed by subtracting the second microphone signal from a delayed version of the first microphone signal.” As discussed for claim 13, Brandstein discloses this limitation or at least renders it obvious. (*Supra* §VI.A.13; Ex. 1002 ¶171.)

Claim 18 would have been obvious.

#### **19. Claim 19**

Claim 19 depends from claim 15 and further recites “wherein the second virtual microphone is formed by subtracting the first microphone signal from a delayed version of the second microphone signal.” Similar to claim 14 relative to claim 13, the analysis for claim 19 is the same as it is for claim 18, with labels switched for the first and second virtual microphones and for the first and second microphone

signals; and as discussed in connection with claim 14, Brandstein discloses the limitation added by claim 19 or at least renders it obvious. (*Supra* §VI.A.14; Ex. 1002 ¶172.) Claim 19 would have been obvious.

## **20. Claim 20**

Claim 20 depends from claim 15 and further recites “wherein the first and second physical microphones are included in a microphone array.” Brandstein discloses this limitation. (*Supra* §VI.A.8; Ex. 1002 ¶173.) Claim 20 would have been obvious.

## **B. Ground 2: Brandstein, Gannot, and Griffiths-Jim**

Claims 4, 13, 14, 18, and 19 all recite limitations relating to delaying a microphone signal. In Ground 1, Petitioner relies on Brandstein’s disclosure that the GSC’s blocking matrix has a “delay-and-subtract” or “delay-and-sum” structure and that the purpose of the delay in delay-and-subtract (or -sum) structures is to time-align signals. (*E.g., supra* §§VI.A.4, VI.A.13.) While Brandstein’s Figure 5.1 does not explicitly label any blocks connected to or part of the blocking matrix as delay elements, Brandstein implies delays for time-alignment are necessary when the target signal is expected to arrive at an angle that is not perpendicular to the array surface. (*Id.*; Ex. 1002 ¶116.) In Ground 1, Petitioner further explains that Griffiths-Jim expressly shows these delays for time-aligning the signals and that such a technique would have been well known to a POSITA and immediately recognized as

included in Brandstein’s teaching when it discusses the delay-and-subtract structure of the blocking matrix. (Ex. 1002 ¶117.)

In Ground 2, Petitioner submits that the delay limitations of claims 4, 13, 14, 18, and 19 would have alternatively been obvious to a POSITA in further view of Griffiths-Jim, which expressly discloses those limitations. (*See supra* §VI.A.4 (discussing how “elements labeled  $\tau_1, \dots, \tau_M$  are delays used to time-align the sensor signals”); Ex. 1002 ¶175.) A POSITA would have been motivated to combine Griffiths-Jim’s teachings with Brandstein because Brandstein discusses the same original GSC technique that Griffiths-Jim does, just with different notation and different choices on which details to emphasize. (Ex. 1002 ¶176.) Brandstein teaches that the “simplest structure for the BM [i.e., blocking matrix] is a delay-and-subtract beamformer” and that, “[a]ssuming a look direction perpendicular to the array surface, no delay element is necessary.” (Ex. 1003, 88.) As Brandstein itself suggests, an ordinarily skilled artisan would have understood that, for some applications, the target signal would arrive at an angle not perpendicular to the array surface and thus necessitate delay elements, as shown in Griffiths-Jim, which does not assume a perpendicular look direction. (Ex. 1002, ¶176.) Brandstein’s discussion of Figure 5.1 is based on, and cites, Griffiths-Jim. (*E.g.*, Ex. 1003, 88 (citing reference [27]), 108 (reference [27] is Griffiths-Jim).) Combining Griffiths-Jim’s teaching of delay elements  $\tau_1, \dots, \tau_M$  to time-align the sensor signals (*see* Ex. 1005, 27 & Fig. 4) with

Brandstein’s teachings would have merely involved applying a known technique to a similar device (if not the identical device) in the same way with predictable results. (Ex. 1002 ¶176.) A POSITA would have had a reasonable expectation of success of combining the teachings of Griffiths-Jim and Brandstein because they describe the same solution; and for the same reason, adding Griffiths-Jim to the combination does not interfere with the motivation to combine the original GSC (Brandstein) with Gannot or the expectation of success in doing so as discussed in Ground 1. (*Id.*)

Thus, claims 4, 13, 14, 18, and 19 would have been obvious in view of Brandstein, Gannot, and Griffiths-Jim. *KSR*, 550 U.S. at 416-17.

**C. Ground 3: Brandstein, Gannot, and McCowan**

Claims 4, 13, 14, 18, and 19 all recite limitations relating to delaying a microphone signal. Claims 5 and 6 depend from claim 4 and further recite limitations relating to raising the delay to a power that relates to the standard near-field design. (*See supra* §§VI.A.5-VI.A.6.)

In Ground 1, Petitioner relies on Brandstein’s disclosure that the GSC’s blocking matrix has a delay-and-sum (or delay-and-subtract) structure because the microphone signals are delayed to be aligned in time, i.e., to compensate for the propagation delays in the arrival of the source signal at each microphone. (*E.g., supra* §§VI.A.4, VI.A.5, VI.A.6, VI.A.13.) Brandstein discloses that delay-and-sum beamforming was first developed on the basis of the far-field assumption, but may

also be extended to near-field beamforming; and that the standard near-field design involves delay terms whose exponents satisfy the requirements of claims 5 and 6. (*See supra* §§VI.A.5-VI.A.6.) In Ground 1, Petitioner further explains that McCowan expressly shows how to adjust the GSC for near-field situations (with the same near-field design that Brandstein discloses and that satisfies claims 5 and 6) and that such a technique would have been well known to a POSITA and immediately recognized as included in Brandstein’s teaching when it discusses the delay-and-subtract structure of the blocking matrix, the need to time-align the microphone signals, and the standard near-field design, including the delays for time alignment under near-field assumptions. (*E.g., supra* §VI.A.5.)

In Ground 3, Petitioner submits that the delay limitations of claims 5 and 6, and the more-general delay limitations of claims 4, 13, 14, 18, and 19, would have alternatively been obvious to a POSITA in further view of McCowan, which expressly discloses those limitations. As Petitioner explains for Ground 1 and claim 5, McCowan expressly teaches a near-field compensation unit that adapts the GSC for near-field conditions. (*Supra* §VI.A.5 (citing Ex. 1002 ¶¶121-26).) This near-field compensation is expressed as a matrix in equation 16 and includes a delay term in the form of an exponential, where the exponent 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. (*Id.*; Ex. 1006, 89-93 (discussing equation 16,

which relies on equation 6 (note the  $\tau_i$  terms in the exponents); see also equation 3 and related discussion); Ex. 1002 ¶¶66-70, 126, 180.) McCowan thus discloses the limitation that claim 5 adds. (Ex. 1002 ¶180.) And for the same reasons as discussed in Ground 1, it would have been routine for a POSITA to express the near-field compensation in terms of samples. (*Id.*; see also *supra* §VI.A.6.)

A POSITA would have been motivated to combine McCowan’s teachings with Brandstein because McCowan starts with the same original GSC technique that Brandstein discloses and adapts it for near-field scenarios, which Brandstein expressly contemplates. (Ex. 1002 ¶181.) McCowan teaches that the original GSC does not perform well for near-field target signals because, without compensating for near-field effects, the GSC’s lower path will not totally block the target signal, leading to target-signal cancellation in the output. (Ex. 1006, 93-94.) McCowan expressly teaches successfully introducing a near-field compensation unit to the original GSC that solves this problem. (*Id.*, 90-96; Ex. 1002 ¶¶66-70, 126, 181.) “Once this near-field compensation has been performed, a standard GSC blocking matrix can be employed to block the desired signal from the adaptive path.” (Ex. 1006, 93; see also *id.*, 95 (“The blocking matrix and adaptive noise canceling filters are taken from the standard GSC technique [citation]” (citing Griffiths-Jim)).) Because McCowan not only expressly proposes modifying the original GSC but explains exactly how to modify the original GSC and reports the successful results of

a real-world implementation, a POSITA would have been motivated to combine the teachings of McCowan and Brandstein and would have had a reasonable expectation of success of doing so. (Ex. 1002 ¶181.) Combining McCowan’s teachings with Brandstein’s would have merely involved applying a known technique to a similar device exactly as McCowan expressly teaches with predictable and verified results. (*Id.*) Furthermore, the near-field compensation of McCowan is compatible with Gannot and does not interfere with the motivation to combine the original GSC (Brandstein) with Gannot or the expectation of success in doing so as discussed in Ground 1. (*Id.*)

In short, claims 4, 5, 6, 13, 14, 18, and 19 would have been obvious in view of Brandstein, Gannot, and McCowan. *KSR*, 550 U.S. at 416-17.

## **VII. 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.

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

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

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

Efficiency, fairness, and the 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*”).

**1. Factor 1: Potential Stay**

PO sued Petitioner for infringement of the ’357 patent and other patents less than four months ago. *Jawbone Innovations, LLC v. Meta Platforms, Inc.*, No. 6:23-cv-00158, Dkt. No. 1 (W.D. Tex. Feb. 28, 2023). Meta answered Jawbone’s complaint on May 30, 2023, but nothing else has occurred in the district court litigation. *Id.*, Dkt. No. 13. There is no case schedule. No scheduling conference date has been set. No discovery has been served.

Meta will soon move to transfer the litigation to Northern District of California. Two other Jawbone cases involving the same patents were previously transferred to NDCA on similar facts, and it is likely that Petitioner’s case will be transferred. *In re Google LLC*, 58 F.4th 1379, 1383 (Fed. Cir. 2023) (issuing a writ of mandamus directing WDTX to transfer Jawbone’s lawsuit against Google to

*Meta Platforms, Inc. v. Jawbone Innovations, LLC*  
IPR Petition – U.S. Patent 11,122,357

NDCA); *Jawbone Innovations, LLC v. Amazon.com, Inc., et al.*, No. 2:21-cv-00435-JRG, Dkt. No. 60 (transferring Jawbone’s lawsuit against Amazon to NDCA).

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 Jawbone cases are stayed pending the outcome of IPRs challenging those patents. No. 3:22-cv-06727-TLT, Dkt. No. 105; No. 3:23-cv-00466-TLT, Dkt. No. 137. Below is a list of already-instituted IPR proceedings challenging all of the patents-at-issue in the district court litigation between Jawbone and Petitioner:

- IPR2023-00279, Paper 10 (P.T.A.B. May 24, 2023) (U.S. Pat. No. 8,321,213 patent, claims 1-13);
- IPR2023-00276, Paper 10 (P.T.A.B. May 24, 2023) (U.S. Pat. No. 8,321,213 patent, claims 14-42);
- IPR2022-00797, Paper 12 (P.T.A.B. Dec. 1, 2022) (U.S. Pat. No. 8,321,213 patent, claims 14-41);
- IPR2023-00285, Paper 10 (P.T.A.B. June 7, 2023) (U.S. Pat. No. 8,326,611, claims 29-44);
- IPR2023-00286, Paper 10 (P.T.A.B. June 7, 2023) (U.S. Pat. No. 8,326,611, claims 1-28);

- IPR2022-01027, Paper 11 (P.T.A.B. Dec. 5, 2022) (U.S. Pat. No. 8,467,543);
- IPR2022-01060, Paper 12 (P.T.A.B. Jan. 3, 2023) (U.S. Pat. No. 8,503,691, claims 1-22, 29-40);
- IPR2022-01061, Paper 12 (P.T.A.B. Jan. 3, 2023) (U.S. Pat. No. 8,503,691, claims 23-28, 41-46);
- IPR2022-01059, Paper 12 (P.T.A.B. Dec. 7, 2022) (U.S. Pat. No. 10,779,080);
- IPR2023-00284, Paper 12 (P.T.A.B. May 15, 2023) (U.S. Pat. No. 10,779,080);
- IPR2022-00623, Paper 12 (P.T.A.B. Oct. 31, 2022) (U.S. Pat. No. 7,246,058);
- IPR2023-00222, Paper 7 (P.T.A.B. May 24, 2023) (U.S. Pat. No. 7,246,058);
- IPR2022-00649, Paper 13 (P.T.A.B. Nov. 31, 2022) (U.S. Pat. No. 8,019,091);
- IPR2023-00253, Paper 9 (P.T.A.B. June 6, 2023) (U.S. Pat. No. 8,019,091);
- IPR2022-01124, Paper 11 (P.T.A.B. Jan. 3, 2023) (U.S. Pat. No. 11,122,357);
- IPR2023-00251, Paper 7 (P.T.A.B. June 1, 2023) (U.S. Pat. No. 11,122,357).

Because IPRs have been instituted against all patents-at-issue, a stay is highly likely. This factor weighs against discretionary denial.

**2. Factor 2: Proximity of Trial to FWD**

No trial date has been set. This factor weighs against discretionary denial. *See, e.g., Global Tel\*Link Corp. v. HLFIP Holding, Inc.*, IPR2021-00444, Paper 14 at 16-19 (P.T.A.B. Jul. 22, 2021) (“The lack of a scheduled trial date . . . typically weighs against exercising discretion to deny the Petition.”); *GAF Materials LLC v. Kirsch Research and Dev., LLC*, IPR2021-00192, Paper 14 at 12-13 (P.T.A.B. May 25, 2021); *Skechers U.S.A., Inc. v. Nike, Inc.*, IPR2021-00160, Paper 10 at 9-11 (P.T.A.B. May 19, 2021).

**3. Factor 3: Investment in Parallel Proceeding**

PO filed its complaint on February 28, 2023, and Petitioner filed its answer on May 30, 2023. The parties have invested very little in the litigation, which has no case schedule. There has been no discovery, no proposed claim constructions exchanged, no claim construction briefs submitted, and no expert reports served. Very little will be invested in the litigation over the next several months. The “remaining investment” outweighs any investments made. *Samsung Elecs. Am. Inc. v. Snik LLC*, IPR2020-01428, Paper 10 at 11 (P.T.A.B. Mar. 9, 2021); *Huawei Techs. Co. v. WSOU Invs., LLC*, IPR2021-00226, Paper 10 at 13-14 (P.T.A.B. June 10, 2021).

**4. Factor 4: Overlapping Issues**

If this IPR is instituted, Petitioner stipulates it will not pursue the grounds identified in this Petition before the district court. This factor weighs against discretionary denial. *Sand Revolution II, LLC v. Cont'l Intermodal Grp.-Trucking LLC*, IPR2019-01393, Paper 24 at 11-12 (P.T.A.B. June 16, 2020) (informative) (“*Sand Revolution*”); *Google LLC v. Jawbone Innovations, LLC*, IPR2022-00630, Paper 10 at 16-17 (P.T.A.B. Sept. 13, 2022).

**5. Factor 5: The Parties**

The parties are the same, but based on the schedule and transfer facts above, it is very unlikely that the parties’ district court case will go to trial before this IPR reaches a final written decision. This factor is neutral. *See Google*, IPR2022-00630, Paper 10 at 14.

**6. Factor 6: Other Circumstances**

To Petitioner’s knowledge, other than Amazon’s petition on the same grounds that Petitioner seeks joining in its accompanying motion, the ’357 patent has never been compared to this combination of references by the PTO, any court, or any jury. The merits are particularly strong, which favors institution. *Fintiv*, IPR2020-00019, Paper 11 at 18.

The *Fintiv* analysis favors institution.

**B. Prior IPR Petitions (*General Plastic*)**

IPR petitions against the patent have been filed by Google (IPR2022-01124, filed June 16, 2022), Samsung and Apple (IPR2022-01321, filed July 27, 2022), and Amazon (IPR2023-00251, filed Nov. 21, 2022). These petitions do not warrant discretionary denial under *General Plastic Indus. Co., Ltd. v. Canon Kabushiki Kaisha*, IPR2016-01357, Paper 19 at 15-16 (P.T.A.B. Sept. 6, 2017) (precedential). Indeed, *General Plastic* is “not particularly relevant” in this situation, i.e., where a different petitioner files a “me-too” or “copycat” petition with a timely motion for joinder. *Central Security Group*, IPR2019-01609, Paper 11 at 8; *Celltrion, Inc. v. Genentech, Inc.*, IPR2018-01019, Paper 11 at 9-11 (P.T.A.B. Oct. 30, 2018).<sup>5</sup>

Even considering *General Plastic*, discretionary denial is unwarranted. Meta and the prior petitioners are not the same party and have no significant relationship. They are not co-defendants. They are competitors accused of infringement—in different actions in different courts—based on sales of different products. This weighs

---

<sup>5</sup> This is Meta’s first IPR petition challenging these claims of the ’357 Patent, and Meta is not time-barred. The reasons cited for the Board’s denial of joinder in *Apple v. Uniloc 2017 LLC*, IPR2020-00854, Paper 9 (P.T.A.B. Oct. 28, 2020) (precedential)—where petitioner had filed a previous petition and sought a second petition through joinder when it was time-barred (*id.* at 4, 7)—are not present.

*Meta Platforms, Inc. v. Jawbone Innovations, LLC*  
IPR Petition – U.S. Patent 11,122,357

against denial. *NetNut Ltd. v. Bright Data Ltd.*, IPR2021-00465, Paper 11 at 9 (P.T.A.B. Aug. 12, 2021) (declining to extend *Gen-eral Plastic* and *Valve* to different petitioner with no relationship to previous petitioners); *Mercedes-Benz USA, LLC v. Carucel Invs. L.P.*, IPR2019-01404, Paper 12 at 11-12 (P.T.A.B. Jan. 22, 2020); *Toshiba Am. Info. Sys., Inc. v. Wallelex Microelecs. Ltd.*, IPR2018-01538, Paper 11 at 20 (P.T.A.B. Mar. 5, 2019).

The remaining factors consider timing. But the sequential IPR filings here were caused by PO’s sequential district court filings and should not be held against Petitioner. PO sued Samsung in May 2021 (2:21-cv-00186-JRG-RSP (E.D. Tex.)). PO waited four months before suing Google and Apple (6:21-cv-00985-ADA, 6:21-cv-00984-ADA (W.D. Tex.)), and then waited another two months to sue Amazon. PO then waited another fourteen months, until February 28, 2023, to sue Meta. The Board should not deny Petitioner’s petition merely because PO sued other defendants earlier. *Toshiba*, IPR2018-01538, Paper 11 at 21-22 (PO’s complaint about multiple IPRs “is not persuasive when the volume appears to be a direct result of its own litigation activity”); *id.* at 22 (not unfair for petitioner to file close to one-year bar date even when co-defendants in related litigation “with distinct interests” and “distinct litigation strategies” filed earlier).

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

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

Though this petition presents the same grounds as the petition filed by Amazon (IPR2023-00251), Petitioner is filing this petition to preserve its ability to pursue a final written decision on these grounds in the event that Amazon terminates its involvement prior to entry of a final written decision in the Amazon IPR. If Amazon does not terminate and Petitioner is joined to the Amazon IPR, Petitioner will take an understudy role in the Amazon IPR. Thus, the proposed grounds will not be cumulative of references previously considered by the Board.

**X. MANDATORY NOTICES**

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

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

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

To the best knowledge of Petitioner, the '357 patent is or has been involved in the following district court litigations and petitions for *inter partes* review:

| <b>Name</b>                                         | <b>Number</b> | <b>Court</b> | <b>Filed</b>   |
|-----------------------------------------------------|---------------|--------------|----------------|
| <i>Jawbone Innovations, LLC v. Amazon.com, Inc.</i> | 5:22-cv-06727 | N.D. Cal.    | Nov. 29, 2021  |
| <i>Jawbone Innovations, LLC v. Amazon.com, Inc.</i> | 2:21-cv-00435 | E.D. Tex.    | Nov. 29, 2021  |
| <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>       | 6:21-cv-00985 | W.D. Tex.    | Sept. 23, 2021 |

*Meta Platforms, Inc. v. Jawbone Innovations, LLC*  
 IPR Petition – U.S. Patent 11,122,357

| <b>Name</b>                                                            | <b>Number</b> | <b>Court</b> | <b>Filed</b>  |
|------------------------------------------------------------------------|---------------|--------------|---------------|
| <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> Review                                | IPR2022-01124 | PTAB         | June 16, 2022 |
| Petition for <i>Inter Partes</i> Review                                | IPR2022-01321 | PTAB         | July 27, 2022 |
| Petition for <i>Inter Partes</i> Review                                | IPR2023-00251 | PTAB         | Nov. 21, 2022 |
| <i>Jawbone Innovations, LLC v. Google LLC</i>                          | 3:23-cv-00466 | N.D. Cal.    | Feb. 1, 2023  |
| <i>Jawbone Innovations, LLC v. Meta Platforms, Inc.</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. Guandong Oppo Mobile Telecom. Corp.</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 |

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

Meta provides the following designation of counsel:

| <b>Lead Counsel</b>           | <b>Backup Counsel</b>           |
|-------------------------------|---------------------------------|
| Lisa Nguyen (Reg. No. 58,018) | David Tennant (Reg. No. 48,362) |

|                                                                                                                                                                                                                                                                   |                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                              |
|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <p><a href="mailto:lisa.nguyen@allenoverly.com">lisa.nguyen@allenoverly.com</a></p> <p><u>Postal and Hand-Delivery Address:</u><br/>Allen &amp; Overy LLP<br/>550 High Street<br/>Palo Alto, CA 94301<br/>Telephone: 650-388-1724<br/>Facsimile: 202-683-3999</p> | <p><a href="mailto:david.tennant@allenoverly.com">david.tennant@allenoverly.com</a><br/>Alan Billharz (Reg. No. 79,532)<br/><a href="mailto:Alan.billharz@allenoverly.com">Alan.billharz@allenoverly.com</a><br/>Colby Davis (Reg. No. 81,575)<br/><a href="mailto:colby.davis@allenoverly.com">colby.davis@allenoverly.com</a></p> <p><u>Postal and Hand-Delivery Address:</u><br/>Allen &amp; Overy LLP<br/>1101 New York Ave. NW<br/>Washington, D.C. 20005<br/>Telephone: (202) 683-3891<br/>Facsimile: (202) 683-3999</p> <p>Eric Lancaster (<i>pro hac vice</i> to be filed)<br/><a href="mailto:eric.lancaster@allenoverly.com">eric.lancaster@allenoverly.com</a></p> <p><u>Postal and Hand-Delivery Address:</u><br/>Allen &amp; Overy LLP<br/>550 High Street<br/>Palo Alto, CA 94301<br/>Telephone: 650-388-1724<br/>Facsimile: 202-683-3999</p> <p>Daniel Margolis (<i>pro hac vice</i> to be filed)<br/><a href="mailto:daniel.margolis@allenoverly.com">daniel.margolis@allenoverly.com</a><br/>Grace Wang (Reg. No. 69,892)<br/><a href="mailto:grace.wang@allenoverly.com">grace.wang@allenoverly.com</a></p> <p><u>Postal and Hand-Delivery Address:</u><br/>Allen &amp; Overy, LLP<br/>1221 6th Ave<br/>New York, NY 10020<br/>Telephone: (212) 756-1143<br/>Facsimile: (202) 683-3999</p> |
|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|

Pursuant to 37 C.F.R. § 42.10(b), a Power of Attorney accompanies this Petition.

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

Service information for counsel is provided above. Meta consents to service by email at the following email address: AO\_Meta\_Jawbone@AllenOvery.com

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

The fee set forth in 37 C.F.R. §42.15(a) for this petition has been paid. The undersigned further authorizes payment for any additional fees that may be due in connection with this petition to be charged to Deposit Account 60-4184.

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

Meta certifies that the '357 patent is available for IPR and that Meta is not barred or estopped from requesting IPR. This petition is being filed within one year of service of the Complaint in the parties' related district court litigation.

**CONCLUSION**

Petitioner respectfully requests cancellation of the challenged claims.

Respectfully submitted,

Allen & Overy, LLP

June 27, 2023

/Lisa Nguyen/

Lisa Nguyen (Reg. No. 58,018)

Counsel for Petitioner,  
META PLATFORMS, INC.

**CERTIFICATE OF COMPLIANCE**

Pursuant to 37 C.F.R. § 42.24(d), the undersigned certifies that foregoing PETITION FOR *INTER PARTES* REVIEW OF CLAIMS 1-20 OF U.S. PATENT NO. 11,122,357 PURSUANT TO 35 U.S.C. §§ 311–319, 37 C.F.R. § 42, exclusive of the parts exempted as provided in 37 C.F.R. § 42.24(a), contains 13,974 words and therefore complies with the type-volume limitations of 37 C.F.R. § 42.24(a).

Allen & Overy, LLP

June 27, 2023

/Lisa Nguyen/  
Lisa Nguyen (Reg. No. 58,018)

Counsel for Petitioner,  
META PLATFORMS, INC.

**CERTIFICATE OF SERVICE**

I hereby certify that a true and correct copy of the foregoing **PETITION FOR *INTER PARTES* REVIEW OF CLAIMS 1-20 OF U.S. PATENT NO. 11,122,357 PURSUANT TO 35 U.S.C. §§ 311–319, 37 C.F.R. § 42 and EXHIBITS 1001-1013** are being served on June 27, 2023, via FedEx Priority Overnight on counsel of record for U.S. Patent No. 11,122,357 as addressed below:

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

A courtesy copy is also being served on counsel via electronic mail for the patent holder in the pending district court litigation, *Jawbone Innovations, LLC v. Meta Platforms, Inc.*, No. 6:23-cv-00158 (W.D. Tex., filed February 28, 2023), and all counsel for the patent owner in IPR2023-00251.

Alfred R. Fabricant  
Peter Lambrianakos  
Vincent J. Rubino, III  
Enrique W. Iturralde  
**FABRICANT LLP**  
411 Theodore Fremd Avenue,  
Suite 206 South  
Rye, New York 10580  
Telephone: (212) 257-5797  
Facsimile: (212) 257-5796  
Email: [ffabricant@fabricantllp.com](mailto:ffabricant@fabricantllp.com)  
Email: [plambrianakos@fabricantllp.com](mailto:plambrianakos@fabricantllp.com)  
Email: [vrubino@fabricantllp.com](mailto:vrubino@fabricantllp.com)

Raymond W. Mort, III  
**THE MORT LAW FIRM, PLLC**  
501 Congress Avenue, Suite 150  
Austin, Texas 78701  
Telephone: (512) 865-7950  
Email: [raymort@austinlaw.com](mailto:raymort@austinlaw.com)

*Meta Platforms, Inc. v. Jawbone Innovations, LLC*  
IPR Petition – U.S. Patent 11,122,357

Email: [jawbone@fabricantip.com](mailto:jawbone@fabricantip.com)  
Email: [rcowell@fabricantllp.com](mailto:rcowell@fabricantllp.com)  
Email: [eiturrealde@fabricantllp.com](mailto:eiturrealde@fabricantllp.com)

Allen & Overy, LLP

June 27, 2023

/Lisa Nguyen/  
Lisa Nguyen (Reg. No. 58,018)

Counsel for Petitioner,  
META PLATFORMS, INC.