

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

GOOGLE LLC,
Petitioner,

v.

SOUNDCLEAR TECHNOLOGIES, LLC,
Patent Owner.

Case No. IPR2025-00345
U.S. Patent No. 9,031,259

PETITION FOR *INTER PARTES* REVIEW

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EXHIBITS

Exhibit	Description
1001	U.S. Patent No. 9,031,259 to Yamabe (“the ’259 patent”)
1002	Prosecution History of U.S. Patent No. 9,031,259
1003	Declaration of Stuart Lipoff (“Lipoff”)
1004	CV of Stuart Lipoff
1005	U.S. Patent Appl. Pub. No. 2011/0026730 to Li et al (“Li-730”)
1006	U.S. Patent Appl. Pub. No. 2012/0027219 to Kale et al. (“Kale”)
1007	U.S. Patent Appl. Pub. No. 2011/0135107 to Konchitsky et al. (“Konchitsky”)
1008	U.S. Patent Appl. Pub. No. 2010/0111329 to Namba et al. (“Namba”)
1009	U.S. Patent Appl. Pub. No. 2012/0197638 to Li et al. (“Li-638”)
1010	U.S. Patent Appl. Publ. No. 2012/0128187 to Yamada et al. (“Yamada”)
1011	U.S. Patent Appl. Pub. No. 2010/0303267 to Pedersen et al. (“Pedersen”)
1012	U.S. Patent Appl. Pub. No. 2012/0022373 to Tateyama (“Tateyama”)
1013	U.S. Patent No. 8,615,392 to Goodwin et al. (“Goodwin”)
1014	<i>SoundClear Technologies LLC v. Google LLC</i> , Case No. 1:24-cv-00729, Complaint for Patent Infringement (E.D. VA May 1, 2024)

I. Introduction

Petitioner, Google LLC, requests *inter partes* review of claims 1-20 of U.S. Patent No. 9,031,259 (“the ’259 patent”) (Ex. 1001), assigned to Patent Owner SoundClear Technologies LLC, and cancellation of the same based on the following grounds:

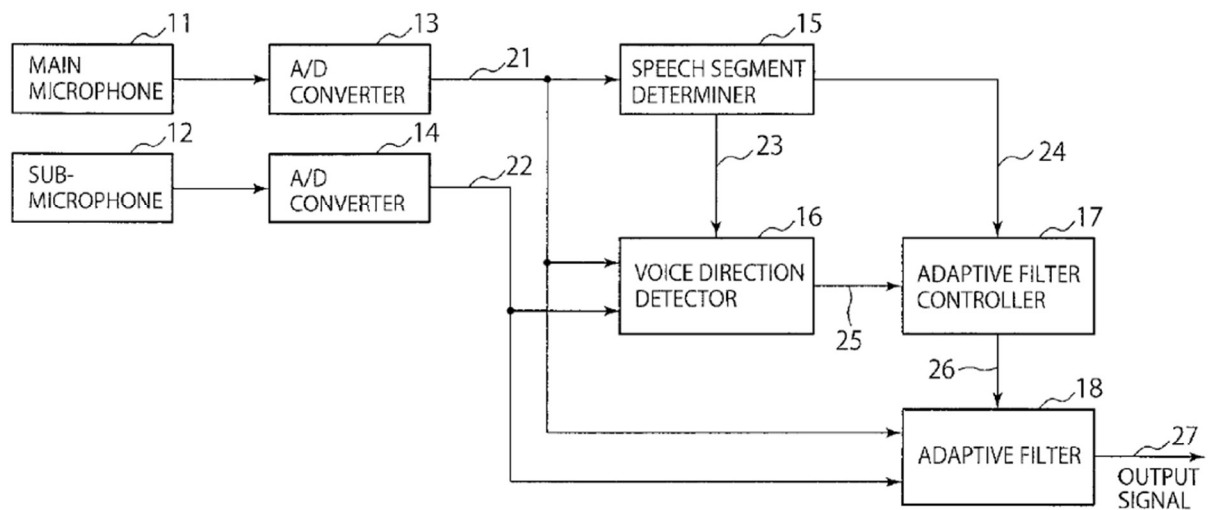
Grounds	Claims	Basis	Prior Art
1	1-3, 5, 6, 8, 13-15, and 17-19	§103	Kale
2	2, 3, 5, 6, 8, 9, 14, 15, and 17-20	§ 103	Kale and Namba
3	2-4, 7, and 14-16	§ 103	Kale and Li-638
4	10-11	§ 103	Kale and Yamada
5	1-3 and 12-15	§103	Li-730

II. The ’259 Patent (Ex. 1001)

The ’259 patent, filed September 14, 2012 and issued May 12, 2015, claims priority to Japanese Patent Application Nos. JP 2011-201759 and JP 2011-201760, both filed September 15, 2011. Ex. 1001, cover.¹

¹ Petitioner evaluates the claims under pre-AIA standards.

The '259 patent generally relates to “a noise reduction apparatus, an audio input apparatus, a wireless communication apparatus, and a noise reduction method.” Ex. 1001, 1:17-19. The noise reduction apparatus includes “main” microphone 11, “sub-microphone” 12, A/D converters 13 and 14, speech segment determiner 15, voice direction detector 16, adaptive filter controller 17, and adaptive filter 18.” Ex. 1001, 3:55-59, FIG. 1; Lipoff, ¶56.



1

FIG. 1

Ex. 1001, FIG. 1.

Sounds received by the main microphone are converted into a first “sound pick-up signal” then input into “speech segment determiner.” Ex. 1001, 4:11-20, FIG. 1. The speech segment determiner “determines whether or not [the] sound picked up the main microphone 11 is a speech segment (voice component)” and

“outputs speech segment information 23 and 24 to the voice direction detector 16 and the adaptive filter controller 17, respectively.” Ex. 1001, 4:40-49, FIG. 1. Sounds received by the sub-microphone are converted into a second “sound pick-up signal” then input into the “voice direction detector.” Ex. 1001, 4:41-26, FIG. 1; Lipoff, ¶57.

The voice direction detector determines “which direction a voice sound travels” using either a “phase difference”² or “magnitude”³ based calculation, and “outputs voice incoming-direction information 25 to the adaptive filter controller 17,” which controls the adaptive filter. 16:67-17:9. The adaptive filter may use the speech segment information, sound pick-up signals, and the “phase difference” (PD1) or “power difference” (magnitude) (PD2) calculated by the

² “Phase difference” relates to the “amount of delay” between first and second sound signals. Ex. 1001, 18:41-45.

³ “[M]agnitude is the maximum amplitude, an integral value of amplitude of the Sound pick-up signals 21 and 22, etc.” Ex. 1001, 25:10-12. “The difference and the ratio between the magnitudes of sounds are referred to as a power difference and a power ratio, respectively. Both factors are referred to as power information.” Ex. 1001, 17:6-10.

voice direction detector to perform noise reduction. Ex. 1001, 1:57-61, 22:38-50, 23:53-24:52, 25:15-34, 25:44-51, 25:62-26:9; Lipoff, ¶¶58-60.

III. Level of Ordinary Skill in the Art

A person of ordinary skill in the art (“POSITA”) in the field of the ’259 patent would have had at least a bachelor’s degree in electrical engineering or related discipline, and at least two years of experience in voice or audio processing. Relevant work experience can substitute for formal education and additional education could substitute for work experience. Lipoff, ¶61.

IV. Claim Construction

The Board construes claims using the standards set forth in *Phillips v. AWH Corp.*, 415 F.3d 1303 (Fed. Cir. 2005) (en banc). Claims should be construed only to the extent necessary to resolve a controversy. *Nidec Motor Corp. v. Zhongshan Broad Ocean Motor Co.*, 868 F.3d 1013, 1017 (Fed. Cir. 2017). No terms need construction to resolve the unpatentability issues in this Petition.⁴

⁴ Petitioner may raise indefiniteness and/or claim constructions in other forums.

Such challenges do not preclude the Board’s ability to resolve patentability.

V. Ground 1: Claims 1-3, 5, 6, 8, 13-15, and 17-19 Are Obvious over Kale

A. Kale (Ex. 1006)

Kale, filed July 28, 2010, and published February 2, 2012, qualifies as prior art to the '259 patent under at least §102(e). Kale discloses a “noise cancellation device” and method. Kale, Abstract. The device includes, *inter alia*, two microphones, computation/FFT modules, a direction of arrival (DOA) module, and a beamformer. Kale, [0028], FIG. 8.

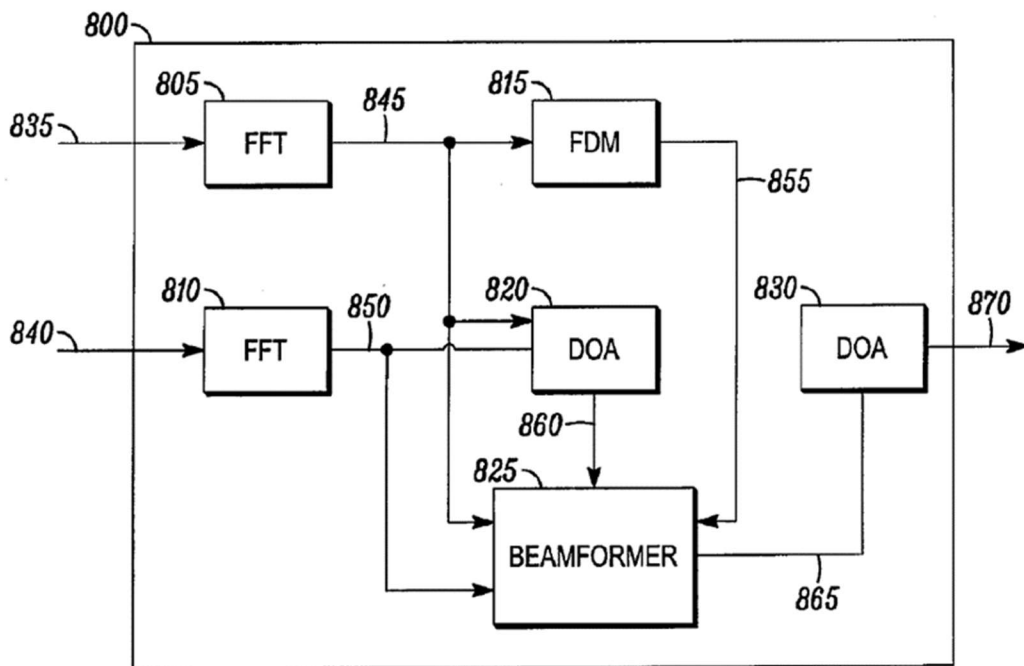


FIG. 8

Kale, FIG. 8.

Kale’s microphones detect “two [sound] sources” (e.g., “speech” and “noise”) and “generate respective raw audio data 835, 845.” Kale, [0028] [0043],

FIGS. 8-9. The computation/FFT modules “determine which of the raw audio data signals comprises the speech signal” using “factors ... such as common formants, formants with known patterns, etc.” and output a “speech signal” 845, 850 to the DOA module and beamformer. Kale, [0003], [0043]-[0044], FIG. 8. The DOA module “determin[es] the direction from which the speech data arrived” and outputs a “DOA data signal” 860 containing the direction information to the beamformer. Kale, [0028], [0031], [0046], FIGS. 8-9. Finally, the beamformer uses the inputted speech signal and direction information to “modify [the] audio signal to isolate a speech therein” or “enhance a speech of an audio signal” by “eliminating” the noise contained in the received audio signal.” Kale, [0040], [0047]-[0048], FIGS. 8-9.

Kale is analogous art to the '259 patent because it is in the same field of endeavor as the '259 patent: audio signal processing. *Compare* Kale, Abstract, *with* Ex. 1001, 1:17-19; Lipoff, ¶73. Furthermore, Kale is directed to the same problem as the '259 patent: improving noise reduction methods to enhance voice sound quality. *Compare* Kale, [0016], [0022], *with* Ex. 1001, 1:38-42; Lipoff, ¶73.

B. Independent Claims

Kale at least suggests each element of the independent claims because it discloses the following (Lipoff, ¶73):

1. Claim 1

a. [1preamble]: “A noise reduction apparatus comprising:”

Kale discloses [1preamble]⁵ because it discloses a “noise cancellation device” Kale, [0003], [0028], FIG. 8; Lipoff, ¶83. Figure 8 below illustrates the noise cancellation device including *inter alia* a “beamformer” configured to “modify an audio signal to isolate a speech therein” or “enhance a speech of an audio signal” by “eliminating” the noise contained in the received audio signal. Kale, Abstract, [0003], [0023], [0034], [0047], FIG. 8.

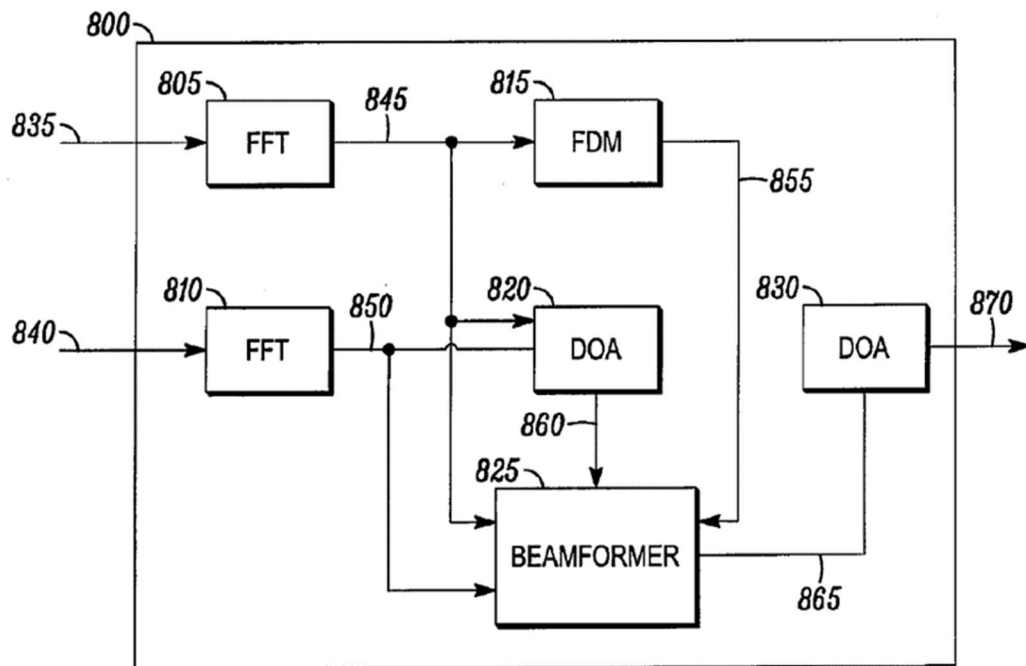


FIG. 8

⁵ Petitioner does not represent that the preamble is limiting.

Kale, FIG. 8.

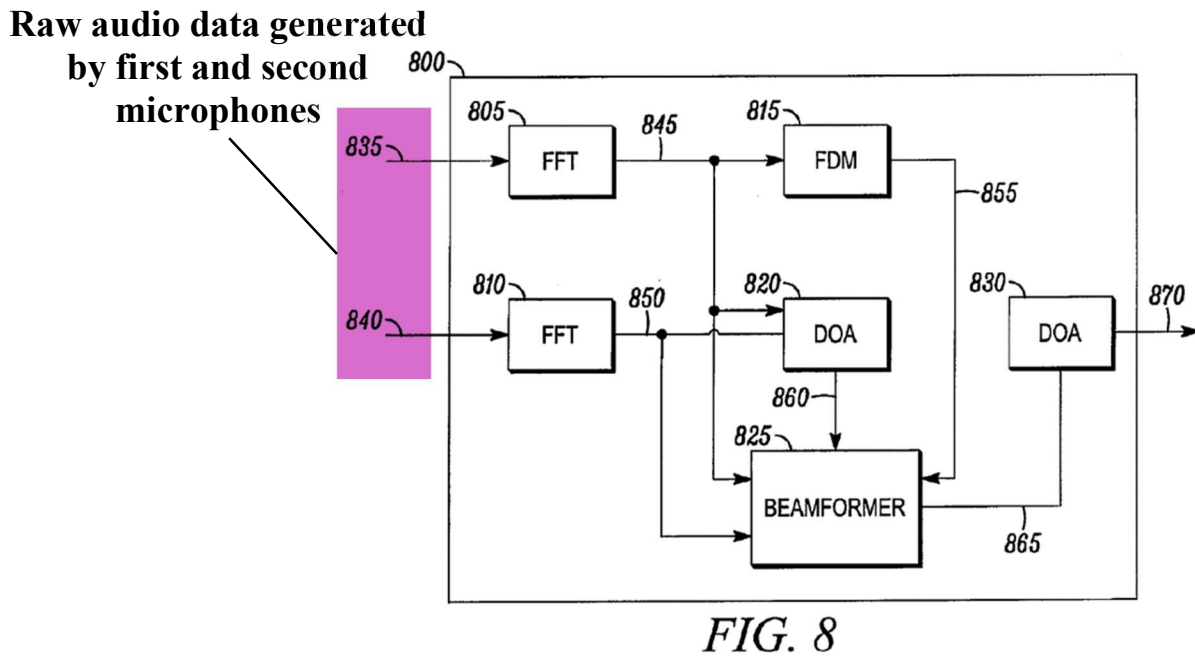
- b. [1a]: “a speech segment determiner configured to determine whether or not a sound picked up by at least either a first microphone or a second microphone is a speech segment and”

Kale discloses [1a] because the noise cancellation device includes computation modules (a speech segment determiner), e.g., Fast Fourier Transform (FFT) modules 805, 810, for receiving the raw audio data and generating “speech data” signals (845, 850) therefrom. Kale, [0003], [0029]-[0031], FIG. 8, claims 1 and 3⁶; Lipoff, ¶¶84-88.

Kale discloses at least either a first or second microphone for picking up sound. Kale, [0028], [0043], FIGS. 8-9; Lipoff, ¶84.

As illustrated in annotated Figure 8 below, Kale’s noise cancellation device includes “two microphones” for detecting “two [sound] sources” (e.g., “speech” and “noise”) and “generat[ing] respective raw audio data 835, 845.” Kale, [0028], [0043], FIGS. 8-9.

⁶ Published claims 1 and 3 were first presented in the as-filed application on July 28, 2010.



Kale, FIG. 8 (annotated).

Kale discloses a computation module (speech segment determiner) for “determin[ing] which of the raw audio data signals comprises the speech signal.” Kale, [0003], [0029]-[0031], [0044], FIG. 8, claims 1 and 3; Lipoff, ¶¶86-88.

The computation module, e.g., FFT module, “determine[s] which of the raw audio data signals comprises the speech signal” using “factors ... such as common formants, formants with known patterns, etc.” Kale, [0043]-[0044]; *see also* [0003] (“A first transformed signal relates to speech data”), claim 1 (“first computation modules receiv[e] raw audio data and generat[e] a respective transformed signal as a function of formants, a first transformed signal relating to speech data), claim 3

(“wherein the first computation modules are Fast Fourier Transform (FFT) modules.”). This is similar to the speech segment determiner of the ’259 patent which also uses formants to detect speech segments. Ex. 1001, 5:33-36 (“[s]pectra (referred to as formant, hereinafter) represent the feature of a voice and are to be detected in determining speech segments by the speech determination unit 38.”).

Kale further discloses that “[t]he exemplary embodiments estimate[] formant position and/or maximum speech energy regions in real time using formant tracking algorithms such as Linear Predictive Coding (LPC), Hidden Markov Model (HMM), etc.,” all of which are known speech detection techniques. Kale, [0024]. Thus, Kale’s **computation/FFT module** is a speech segment determiner configured to determine whether or not **a sound picked up** (raw audio data 835) by at least either a first microphone or a second microphone is a speech segment.

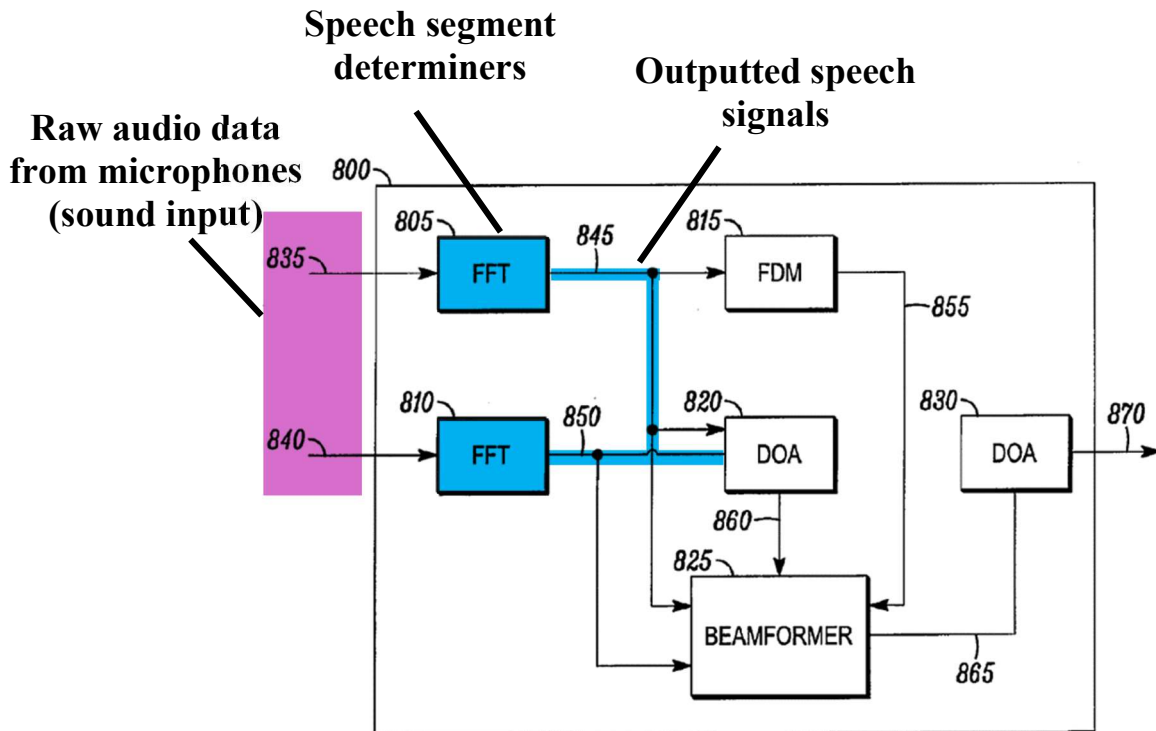
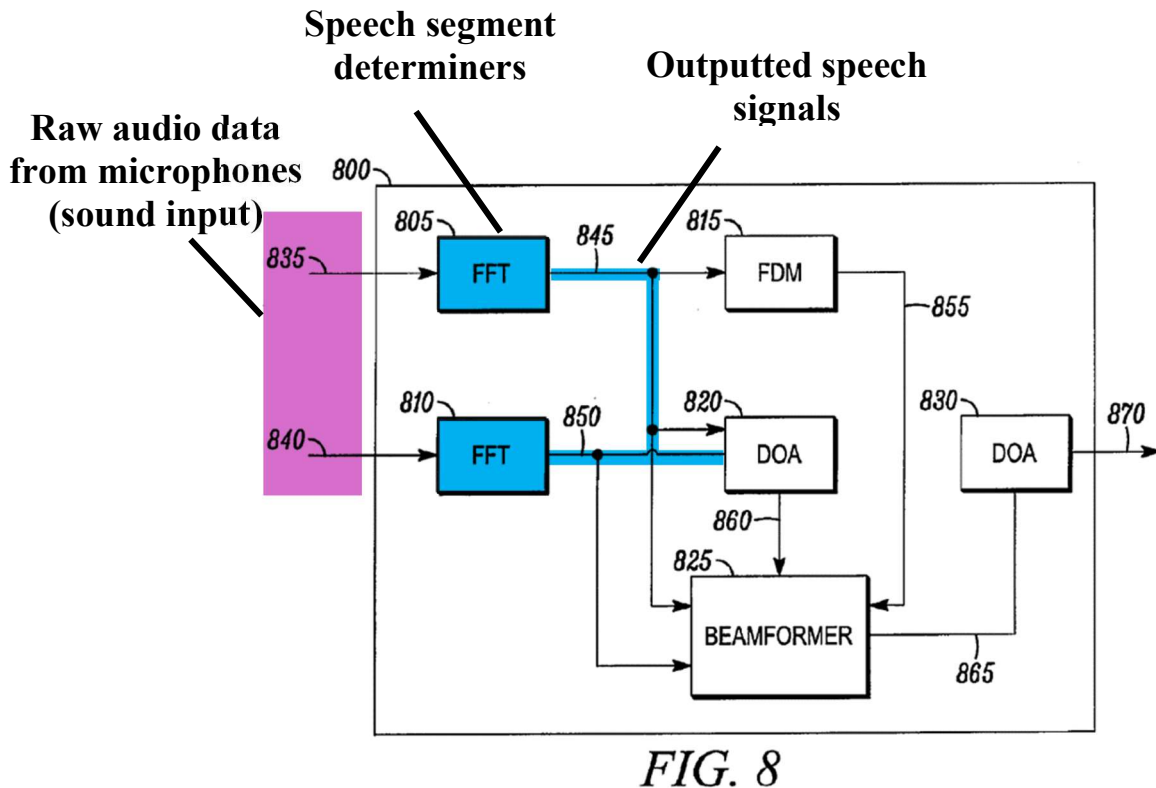


FIG. 8

Kale, FIG. 8 (annotated).

- c. [1b]: “to output speech segment information when it is determined that the sound picked up by the first or the second microphone is the speech segment;”

Kale discloses [1b] because, as explained for [1a], the FFT module (speech segment determiner) outputs “speech data” signals 845, 850 (speech segment information) “[u]pon determining which microphone received the speech signal.” Kale, [0029], [0044], FIG. 8; Lipoff, ¶¶89-90. The speech segment information (contained in the speech signal) is received by a “direction of arrival module” and “beamformer” as discussed in [1c] and [1d], *infra*. Kale, [0044].



Kale, FIG. 8 (annotated).

- d. [1c]: “a voice direction detector configured, when receiving the speech segment information, to detect a voice incoming direction indicating from which direction a voice sound travels,”

Kale discloses [1c] because the noise cancelling device includes a “**direction of arrival (DOA) module**” for “determining the direction from which the speech data arrived.” Kale, [0028], [0031], [0046], FIGS. 8-9; Lipoff, ¶¶91-92. Annotated Figure 8 below illustrates “[t]he **direction of arrival module** receiv[ing] the **first and second transformed signals**” (**speech segment information**). Kale, [0003], [0046], FIG. 8.

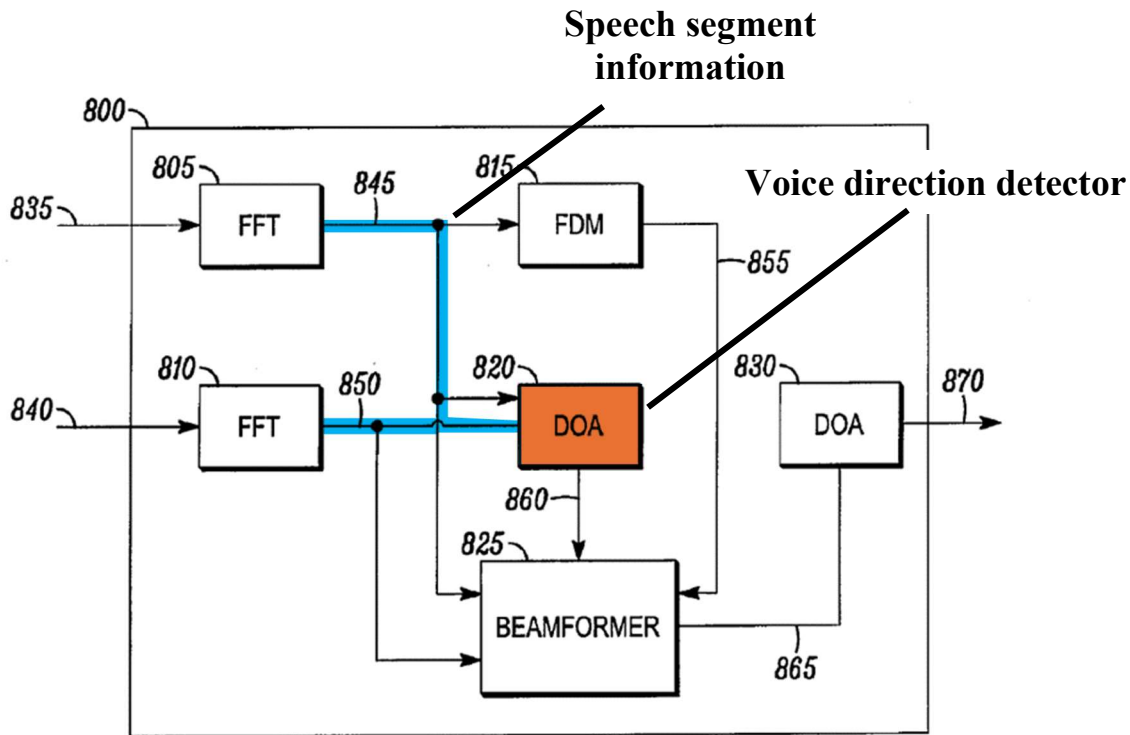


FIG. 8

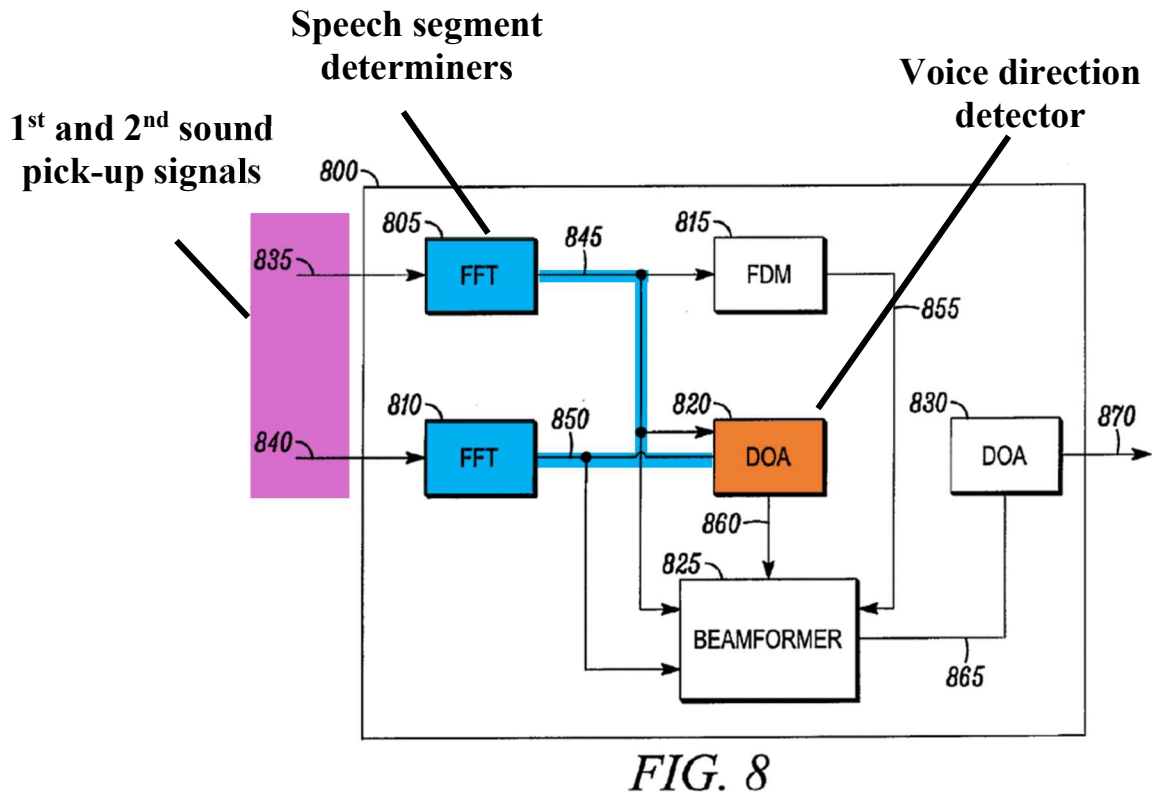
Kale, FIG. 8 (annotated).

The DOA module calculates the direction of arrival by “determin[ing] a cross-correlation between the first and second transformed signals, and generat[ing] a spatial orientation data signal.” Kale, [003]. More specifically, it calculates the “direction of arrival” of the speech segment by “comput[ing] a cross-correlation between the two [speech and noise] signals 845, 850” to “generate[] a spatial orientation data signal” including “a first angle relating to the speech data and a second angle relating to the noise data.” Kale, [0003], [0031], claim 8. This calculation, as was well understood in the art, indicates the direction at which the speech sound arrived at the first or second microphone. Lipoff, ¶92 (citing

Goodwin, 8:15-26, 8:31-47). Thus, like the '259 patent, which also explains that “a voice incoming direction is detected by... measuring the correlation between sounds picked up by the main microphone 11 and the sub-microphone 12,” Kale discloses a voice direction detector. Ex. 1001, 19:3-9.

- e. **[1c.i]: “based on a first sound pick-up signal obtained based on a sound picked up by the first microphone and a second sound pick-up signal obtained based on a sound picked up by the second microphone”**

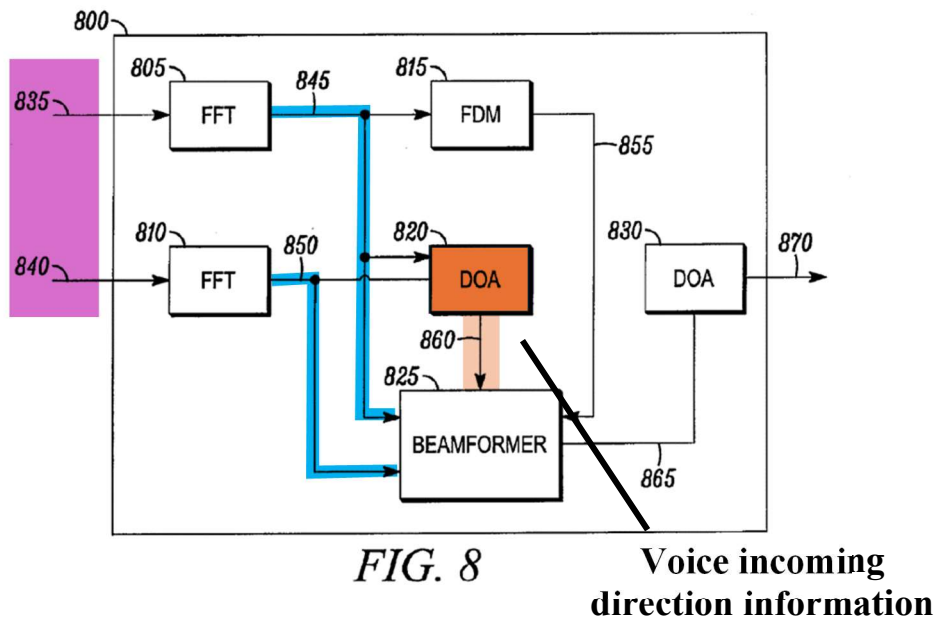
Kale discloses [1c.i] because as explained for [1c] and [1a], the cross-correlation (voice direction determination) is based on first and second “speech signals” 845 and 850 corresponding to first and second raw audio data inputs 835 and 840 (sound pick-up signals) picked up by the first and second microphones. Kale, [0003], [0031], [0043]-[0044], claim 8; Lipoff, ¶93. Annotated Figure 8 below illustrates that “the first and second microphone speech signals 845 and 850” (speech segment information) are based on “raw audio data” 835, 840 (first and second sound pick-up signals) picked up by the first and second microphones. Kale, [0031], [0043]-[0044], FIG. 8.



Kale, FIG. 8 (annotated).

- f. [1c.ii]: “and to output voice incoming-direction information when the voice incoming direction is detected; and”

Kale discloses [1c.ii] because the DOA module (voice direction detector) outputs a “DOA data signal” (voice incoming-direction information) when the speech is detected. Kale, [0031], [0046], Fig. 8; Lipoff, ¶94. The DOA data signal is received by beamformer 825, which uses the signal to “place[] a null at the noise frequency direction for the formant range of frequencies, thereby eliminating the maximum noise in the range.” Kale, [0033], [0034], [0046]- [0047], FIG. 8.

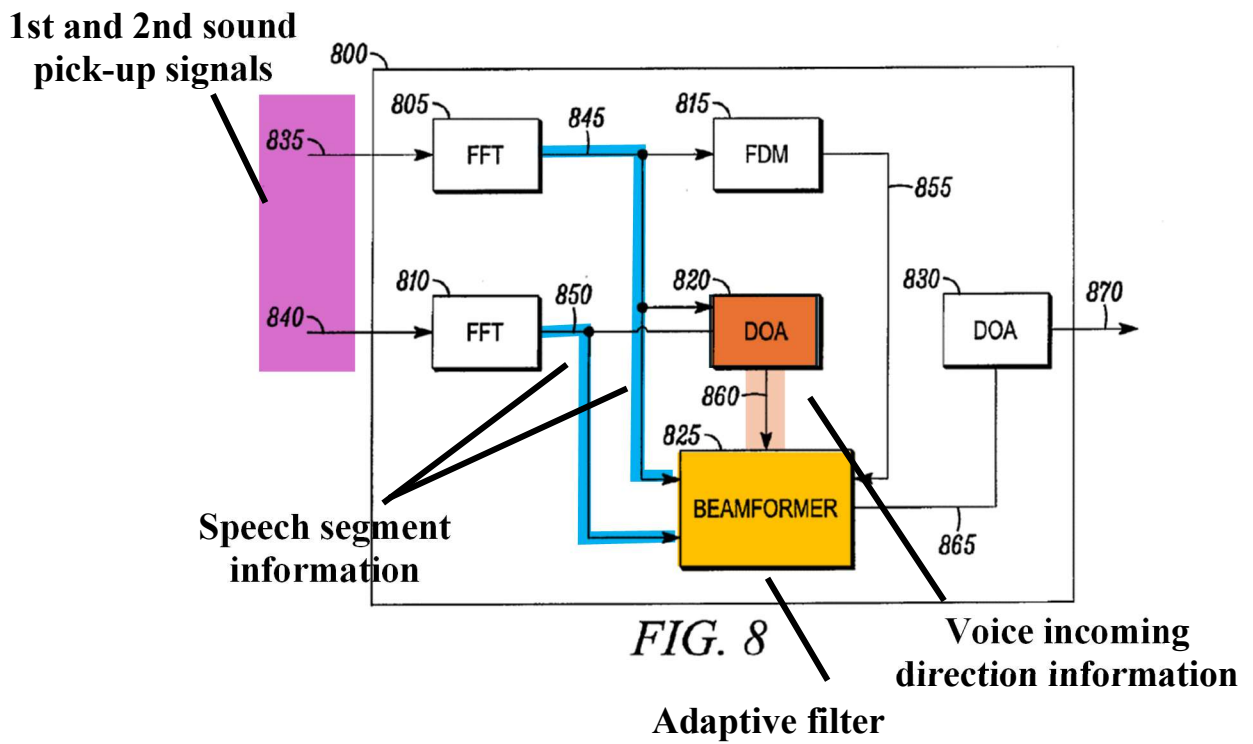


Kale, FIG. 8 (annotated).

- g. [1d]: “an adaptive filter configured to perform a noise reduction process using the first and second sound pick-up signals based on the speech segment information and the voice incoming-direction information.”

Kale discloses [1d] because the noise canceling device includes a “beamformer” (adaptive filter) configured to “modify an audio signal to isolate a speech therein” or “enhance a speech of an audio signal” by “eliminating” the noise contained in the received audio signal” (perform a noise reduction process) (1) using the microphone sound pick-up signals, (2) based on the first and second “microphone speech signal[s]” (speech segment information), and (3) based on “the DOA data signal 860” (voice incoming-direction information). Kale, [0040],

[0047]-[0048], FIGS. 8 and 9; Lipoff, ¶¶95-96. Annotated Figure 8 below illustrates the beamformer receiving the DOA data signal, as well as the first and second speech signals 845, 850 that are generated using the raw audio data signals 835, 840 (first and second sound pick-up signals).



Kale, FIG. 8 (annotated).

2. Claim 13

- a. [13preamble]: "A noise reduction method comprising the steps of:"

Kale discloses [13preamble] because it discloses a "method for noise cancellation using multiple microphones that is formant aided." Kale, [0016], FIG. 9; Lipoff, ¶117.

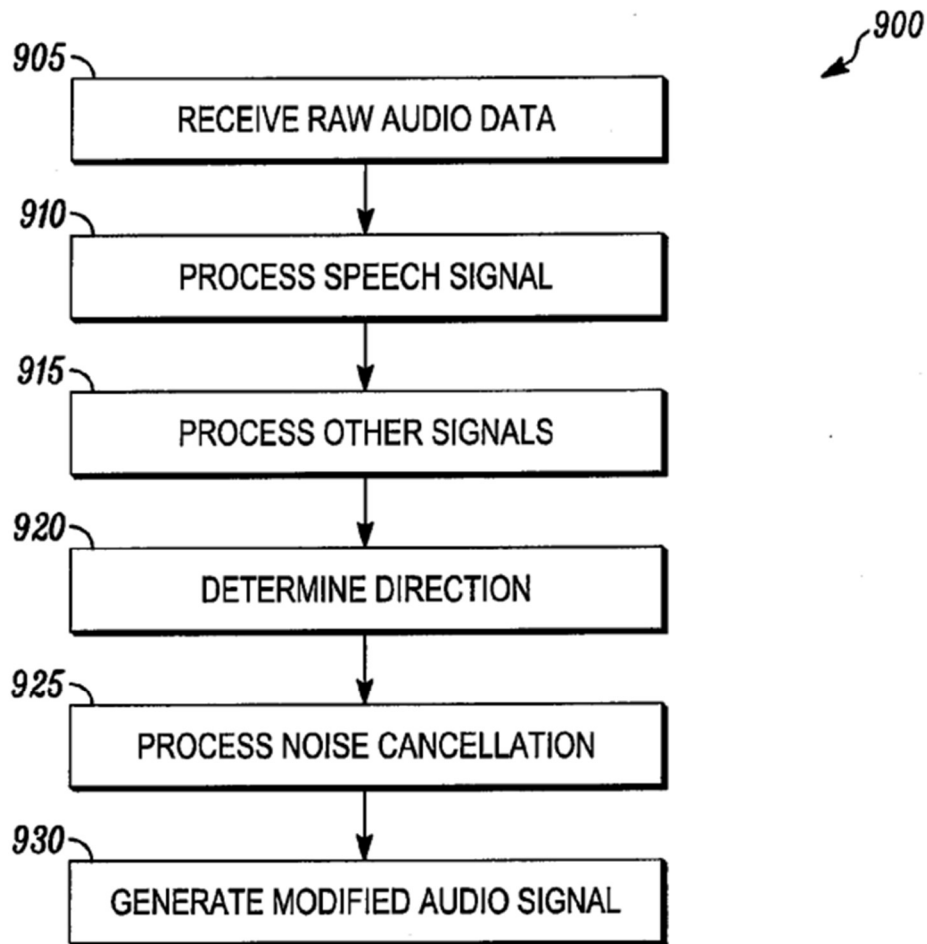


FIG. 9

Kale, FIG. 9.

- b. [13a]: “determining whether or not a sound picked up by at least either a first microphone or a second microphone is a speech segment;”

Kale discloses [13a] for reasons discussed for [1a], *supra*. Lipoff, ¶118.

- c. **[13b]: “detecting a voice incoming direction indicating from which direction a voice sound travels, based on a first sound pick-up signal obtained based on a sound picked up by the first microphone and a second sound pick-up signal obtained based on a sound picked up by the second microphone, when it is determined that the sound picked up by the first or the second microphone is the speech segment; and”**

Claim [13b] incorporates [1b], [1c], and [1c.i]. Kale discloses [13b] for reasons discussed for [1b], [1c], and [1c.i], *supra*. Lipoff, ¶119.

- d. **[13c]: “performing a noise reduction process using the first and second sound pick-up signals based on speech segment information indicating that the sound picked up by the first or the second microphone is the speech segment and voice incoming-direction information indicating the voice incoming direction.”**

Kale discloses [13c] for reasons discussed for [1d], *supra*. Lipoff, ¶120.

C. Dependent Claims

- 1. **Claims 2 and [14]: “the voice direction detector detects the voice incoming direction [voice incoming direction is detected] based on a phase difference between the first and second sound pick-up signals.”**

Kale at least suggests claims 2 and 14 because, as explained for [1c], the DOA module (voice direction detector) detects the arrival direction of the speech by “comput[ing] a cross-correlation between the two [sound] signals 845, 850” and determining a “spatial orientation” of the signals. Kale, [0003], [0031]; Lipoff, ¶97-99.

A POSITA would have understood that Kale’s “cross correlation” pertains to a phase-difference calculation, i.e., the difference between the angle at which the first and second sound pick-up signals are received with respect to a reference microphone. Lipoff, ¶98 (citing Goodwin, 8:15-26, 8:31-47). This is consistent with the ’259 patent, which also explains that “[t]he phase difference of sounds between the main microphone 11 and sub-microphone 12 can be calculated using the cross correlation function.” Ex. 1001, 19:9-12.

Moreover, Kale explains that the DOA module’s output includes the phases of the sound input signals, e.g., “two angles in degrees, the first being for a desired speech signal while the second is for noise.” Kale, [0031]. Output “ θ_N is the direction of noise and [output] θ_S is the direction of sound.” Kale, [0039]. A POSITA would have understood that these angles correspond to the phases of the first and second sound pick-up signals—that is, θ_N corresponds to the second sound pick-up signal when it contains the noise signal and θ_S corresponds to the first sound pick-up signal when it contains the voice signal. Thus, Kale discloses that its DOA module detects the voice incoming direction based on a phase difference between the first and second sound pick-up signals. Lipoff, ¶99.

2. **Claims 3 and [15]: “the adaptive filter performs the noise reduction process [the noise reduction process is performed] to reduce a noise component carried by the first sound pick-up signal using the second sound pick-up signal when the first sound pick-up signal has a more advanced phase than the second sound pick-up signal**

whereas the adaptive filter performs the noise reduction process [the noise reduction process is performed] to reduce a noise component carried by the second sound pick-up signal using the first sound pick-up signal when the second sound pick-up signal has a more advanced phase than the first sound pick-up signal.”

Kale discloses claims 3 and 15 because the beamformer (adaptive noise filter) reduces the noise component of whichever signal contains the advanced speech signal (either the first or second sound pick-up signal) using the delayed noise signal (the other of the first or second sound pick-up signal). Kale, [0034], [0039], Lipoff, ¶¶100-106, 122.

As explained for [1d], Kale’s adaptive filter is configured to “modify an audio signal to isolate a speech therein” or “enhance a speech of an audio signal” by “eliminating” the noise contained in the received audio signal” based on the “first microphone speech signal 845, the second microphone speech signal 850” (speech segment information) and based on the DOA data signal 860” (voice incoming-direction information). Kale, [0047]-[0048], FIGS. 8 and 9.

And, as explained for claim 2, Kale’s DOA module outputs the phases of each sound input signal as “two angles in degrees, the first being for a desired

speech signal while the second is for noise.” Kale, [0031], [0039] (“ θ_N is the direction of noise and θ_S is the direction of sound”). Thus, a POSITA would have understood that the DOA data signal includes data indicating which sound pick-up signal includes an advanced phase because the angles indicate which signal was delayed. Lipoff, ¶102. Indeed, a POSITA would have understood that a small angle corresponds to a signal having an advanced phase, while a large angle indicates that the signal was delayed. Lipoff, ¶102 (citing Li-638, [0030], [0053], [0061]).

Kale also explains that “[t]he beamformer [] places a null *at the noise frequency direction* for the formant range of frequencies, thereby eliminating the maximum noise in the range.” Kale, [0034] (emphasis added). Accordingly, when “ θ_N is the direction of noise and θ_S is the direction of sound ... the requirement is to *place a null at θ_N* and unity at θ_S .” Kale, [0039] (emphasis added). That is, when θ_S (corresponding to the speech signal with the smaller angle of incidence) has an advanced phase relative to θ_N (corresponding to the noise signal), the noise signal is used for noise reduction. Lipoff, ¶103.

Thus, a POSITA would have understood that Kale’s “beamformer” reduces a noise component carried by the first sound pick-up signal using the second sound pick-up signal when the first sound pick-up signal received by the first microphone has a more advanced phase (θ_S) than the second sound pick-up signal received by the second microphone (corresponding to θ_N).

Kale further discloses that either the first or second signal (first sound pick-up signal 845 or second sound pick-up signal 850) may contain the speech signal (corresponding to θ_S) from which noise is reduced. Kale, [0031]-[0032] (explaining that while “signals 845, 850 are assumed to be speech and noise, respectively...the options may be changed so that the first signal 845 represents noise while the second signal 850 represents speech”).

Thus, a POSITA would have also understood that Kale’s “beamformer” reduces a noise component carried by the second sound pick-up signal (speech signal 850) using the first sound pick-up signal (noise signal 845) when the second sound pick-up signal has a more advanced phase (θ_S) than the first sound pick-up signal (corresponding to θ_N). Lipoff, ¶104.

3. Claims 5 and [17]: “wherein the voice direction detector detects the voice incoming direction [the voice incoming direction is detected] based on magnitudes of the first and second sound pick-up signals.”

Kale at least suggests claims 5 and 17 because the DOA module (voice direction detector) determines the arrival direction (“spatial orientation”) of the speech by determining which “peak” in the noise signal or speech signal is more “prominent” (has a greater magnitude). Kale, [0003], [0031]; Lipoff, ¶107-111, 123.

As explained for [1c], the DOA module calculates the direction of arrival by “determin[ing] a cross-correlation between the first and second transformed signals, and generat[ing] a spatial orientation data signal.” Kale, [0003]. For example, the DOA module calculates the “direction of arrival” of the speech segment by “comput[ing] a cross-correlation between the two [speech and noise] signals 845, 850” to “generate[] a spatial orientation data signal” including “a first angle relating to the speech data and a second angle relating to the noise data.” Kale, [0003], [0031], claim 8.

Kale further explains that the DOA module ascertains the “prominen[ce]” of the “peaks” (magnitudes) associated with each of the first and second spatial orientation signals when determining the direction of arrival. Kale, [0031]. A POSITA would have understood that peak prominence (magnitude) is indicative of speech sound proximity and direction as was well known in the art. Lipoff, ¶109, (citing Li-730, [0023] (“As is well known in the art, a substantially flat curve in the spatial spectrum SS is caused by far field noises, and sharp and *dominant peaks in the spatial spectrum SS is caused by near field sounds of a speaker’s voice* and spot noises from the environment.” (emphasis added))); Laroche, 4:57-61 (“Because the primary microphone 106 is much closer to the user 102 than the secondary microphone 108, the intensity level is higher for the primary microphone 106, resulting in a larger energy level received by the primary

microphone 106 during a speech/voice segment, for example.”); Kale, [0021], FIGS. 3-4 (describing and illustrating a beampattern wherein noise acquired “directly in front of the microphone” produces a prominent peak).

Thus, a POSITA would have understood that a determination by the DOA module that a signal is “prominent” corresponds to a determination that the input is closer, e.g., directly in front of the first microphone; whereas, a determination that the signal is not prominent indicates that the sound source is farther away, i.e., in a direction away from the microphone. Li-730, [0023]; Kale, [0021], FIGS. 3-4.

Lipoff, ¶110.

4. **Claims 6 and [18]: “wherein the adaptive filter performs the noise reduction process [the noise reduction process is performed] to reduce a noise component carried by the first sound pick-up signal using the second sound pick-up signal when the first sound pick-up signal has a greater magnitude than the second sound pick-up signal**

whereas the adaptive filter performs the noise reduction process [the noise reduction process is performed] to reduce a noise component carried by the second sound pick-up signal using the first sound pick-up signal when the second sound pick-up signal has a greater magnitude than the first sound pick-up signal.”

Kale discloses claims 6 and 18 because the beamformer (adaptive noise filter) reduces the noise component of whichever signal contains the more prominent signal (either the first or second sound pick-up signal) using the noise

signal (the other of the first or second sound pick-up signal). Kale, [0034], [0039]; Lipoff, ¶¶112-115, 124.

As explained for [1d], Kale's "beamformer" eliminates the noise contained in the received audio signal (1) using the microphone sound pick-up signals, (2) based on the first and second "microphone speech signal[s]," and (3) based on "the DOA data signal 860." Kale, [0047]-[0048], FIGS. 8 and 9.

Kale further discloses that either the first or second speech signals corresponding to sound pick-up signals 835, 840 may contain the "prominent" speech signal from which noise is reduced. Kale, [0031]-[0032] (explaining that while "signals 845, 850 are assumed to be speech and noise, respectively ... the options may be changed so that the first signal 845 represents noise while the second signal 850 represents speech"). And Kale further explains that "[i]f the DOA 820 determines that the second peak of the second signal 850 is not prominent, a null value is provided." Kale, [0031]; *see also* [0034] ("The beamformer [] places a null *at the noise frequency direction* for the formant range of frequencies, thereby eliminating the maximum noise in the range." (emphasis added)), [0039] ("Assuming θ_N is the direction of noise and θ_S is the direction of sound ... the requirement is to *place a null at θ_N* and unity at θ_S ." (emphasis added)).

Thus, a POSITA would have understood that when the first sound pick-up signal corresponding to the “first microphone speech signal” contains the prominent speech signal, the second sound pick-up signal corresponding to the “second microphone speech signal” is used to reduce the noise component (provide a “null”). Kale, [0031]. Conversely, a POSITA would have understood that when the second sound pick-up signal contains the prominent speech signal, the beamformer reduces the noise contained in the second pick-up signal using the less prominent first sound pick-up signal. Kale, [0031]; Lipoff, ¶115.

5. Claims 8 and [19]: “wherein the voice direction detector detects the voice incoming direction [the voice incoming direction is detected] based on a phase difference between the first and second sound pick-up signals and magnitudes of the first and second sound pick-up signal.”

Claim 8 recites a combination of claims 2 and 5 and claim 19 recites a combination of claims 14 and 17. Kale renders obvious claims 8 and 19 for reasons discussed for claims 2, 5, 14 and 17. Lipoff, ¶¶116, 125.

VI. Ground 2: Claims 2, 3, 5, 6, 8, 9, 14, 15 and 17-20 Are Obvious over Kale and Namba

A. Namba (Ex. 1008)

Namba, filed November 3, 2009, and published May 6, 2010, is prior art to the '259 patent under §102(b). Namba, cover.

Namba teaches a sound processing apparatus including, like Kale, a microphone array for receiving a mixture of “noise sound” and “voice,” a unit for

identifying a voice sound, a distance/direction estimator for determining the direction of the voice, and a unit for suppressing noise in the voice signal. Namba, [0020]-[0023], [0070], table 1. The distance/direction estimator may receive the “volume value sequence of the input sound, spectrum of the input sound and the like” and “has a function, based on the input, as a positional information calculation unit that estimates the sound source of the input sound or positional information such as direction information and distance information of the sound source from which a dominant sound contained in the input sound originates.” Namba, [0070].

Namba is analogous art to the '259 patent because it is in the same field of endeavor as the '259 patent: noise reduction. *Compare* Namba, [0002], [0005]-[0006], *with* Ex. 1001, 1:17-19; Lipoff, ¶76. Furthermore, Namba is directed to the same problem as the '259 patent: improving noise reduction methods to enhance voice sound quality. *Compare* Namba, [0008], *with* Ex. 1001, 1:38-42; Lipoff, ¶76

B. Claims

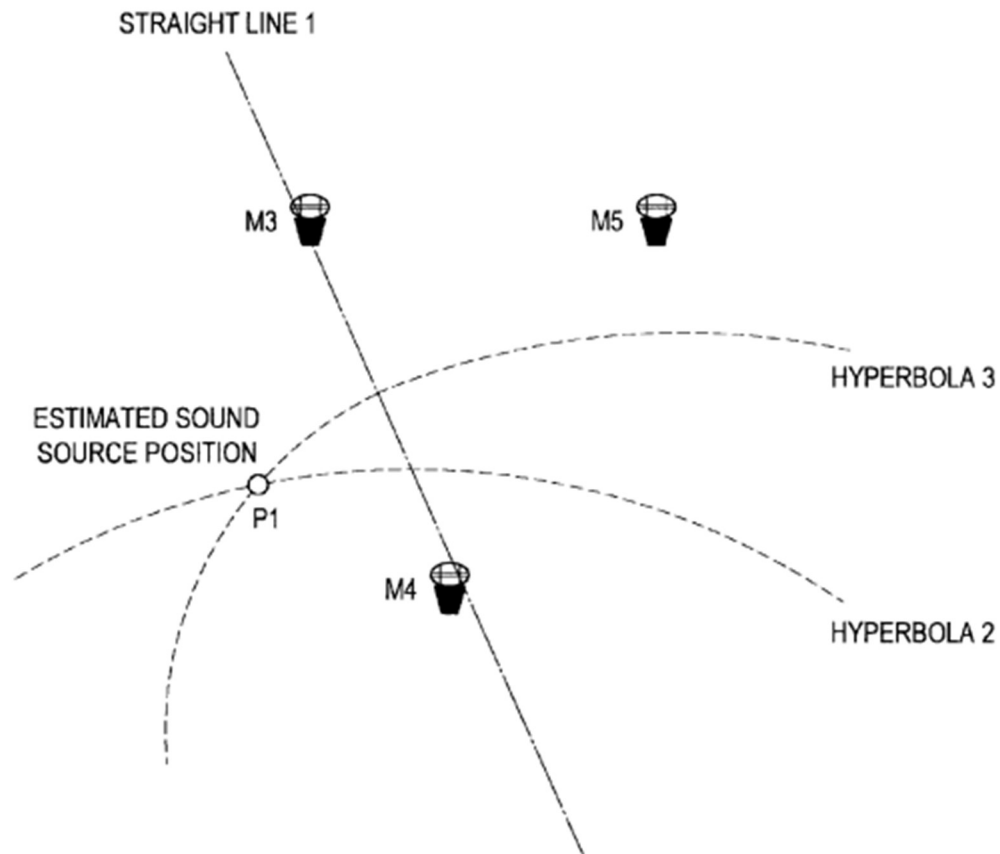
- 1. Claims 2 and [14]: “the voice direction detector detects the voice incoming direction [voice incoming direction is detected] based on a phase difference between the first and second sound pick-up signals.”**

Kale-Namba renders obvious claims 2 and 14 because Namba teaches a sound processing apparatus including a “distance/direction estimator” (voice

direction detector) that determines the direction and distance of a “sound source” “based on a phase difference of two input sounds” of the apparatus. Namba, [0070]-[0073], [0074]-[0076], FIGS. 5 and 6; Lipoff, ¶¶126-129, 144.

The distance/direction estimator measures “the phase of each input sound reaching a microphone,” e.g., “M1 and a microphone M2 ... and a phase difference of the input sounds” and uses the phases and phase difference to calculate “the sound source position of input sound.” Namba, [0071]. For example, “the distance/direction estimator 144 can determine that the sound source is positioned on the opposite side of the microphone M5” when “phase of input sound arriving at the microphone M5 [is] delayed when compared with that of input sound arriving at the microphone M3 or the microphone M4.” Namba, [0076], FIG. 6. The distance/direction determiner can also calculate hyperbolas “on which the sound source could be present based on a phase difference of input sounds arriving at” microphones M3 and M4, or M4 and M5, and “estimate that an intersection P1” of the hyperbolas “is the sound source position.” Namba, [0077], FIG. 6.

FIG.6



Namba, FIG. 6.

a. Motivation to Combine

The techniques implemented by Namba's distance/direction estimator are particularly useful for determining sound input direction because they *combine* phase *and* volume (magnitude) "estimation methods" to detect sound input direction "even if" the sound input signals (sound pick-up signals) experience "reverberation" or "reflection" caused by the sound pick-up apparatus itself.

Namba, [0070]; Lipoff, ¶128. A POSITA would have recognized that sound

signals, as described in Kale, are likewise susceptible to distortions caused by reverberation and reflection (caused e.g., by bouncing off the sound apparatus itself or objects between the sound apparatus and the user's voice) and would have found it beneficial to integrate phase-based voice direction detection, as taught by Namba to reduce such reverberations or reflections that might otherwise reduce the accuracy of the sound input apparatus's voice direction determination. Lipoff, ¶128. Thus, to the extent Patent Owner disputes or the Board finds that Kale does not disclose claim 2, a POSITA would have been motivated to also detect the voice incoming direction based on the phase of the first and second sound pick-up signals to enhance voice direction detection even when the sound input signals experience "reverberation" or "reflection" as taught by Namba. Namba, [0070]; Lipoff, ¶129.

b. Expectation of Success

Kale and Namba use similar elements and techniques to achieve similar results (see Overview of Namba, *supra*) and a POSITA would have had a reasonable expectation of success in configuring Kale's voice direction detector to implement Namba's phase-based voice direction determination process because it would have merely required routine skill in the art to do so. Lipoff, ¶130. For example, a POSITA would have known to use the principles and formulas discussed in [0072]-[0073], [0076] of Namba, taking into account the microphone

position of the noise suppression device of Kale, to ascertain voice incoming direction based on phase and phase difference. Lipoff, ¶130.

2. **Claims 3 and [15]: “the adaptive filter performs the noise reduction process [the noise reduction process is performed] to reduce a noise component carried by the first sound pick-up signal using the second sound pick-up signal when the first sound pick-up signal has a more advanced phase than the second sound pick-up signal**

whereas the adaptive filter performs the noise reduction process [the noise reduction process is performed] to reduce a noise component carried by the second sound pick-up signal using the first sound pick-up signal when the second sound pick-up signal has a more advanced phase than the first sound pick-up signal.”

Kale-Namba renders obvious claims 3 and 15 for the same reasons Kale does, as explained in Ground 1. Lipoff, ¶131, 145. Modifying Kale to incorporate Namba’s phase-based voice direction detection, as described in claims 2 and 14, does not alter this.

3. **Claims 5 and [17]: “wherein the voice direction detector detects the voice incoming direction [the voice incoming direction is detected] based on magnitudes of the first and second sound pick-up signals.”**

Kale-Namba renders obvious claims 5 and 17 because Namba’s “distance/direction estimator” (voice direction detector) determines the direction and distance of a “sound source” “based on a volume difference between input sounds recorded by the microphone[s]” of the apparatus. Namba, [0070], [0074]-[0087], FIGS. 5, 7, 9; Lipoff, ¶¶132-137, 146.

As illustrated in Table 1 below and discussed further below, the distance/direction estimator uses the volume (amplitude) to calculate source sound direction and distance information.

TABLE 1

Block	Input	Output
Volume detector	Input sound	Volume value sequence (amplitude) in frame
Average volume detector	Volume value sequence (amplitude) in frame	Average value of volume
Maximum volume detector	Volume value sequence (amplitude) in frame	Maximum value of volume
Spectrum detector	Input sound	Spectrum
Sound quality detector	Input sound	Likeness of human voice
	Average value of volume	Likeness of music
	Maximum value of volume	Steady or non-steady
	Spectrum	Impulse property
Distance/direction estimator	Input sound	Direction information
	Volume value sequence (amplitude) in frame	Distance information
	Spectrum	

Namba, Table 1 (annotated). Because “magnitude” as described in the ’259 patent includes the “maximum amplitude” of “the sound pick-up signals” and the like (“etc.”), a POSITA would have understood that Namba’s sound source direction determination based on volume/amplitude of the sound input signals is *a voice direction detector that detects voice incoming direction based on magnitude*, as recited in claim 5. Ex. 1001, 25:10-12; Lipoff, ¶133.

Namba further discloses various equations and mathematical principles that can be used to ascertain sound direction and position based on the volume of the input sounds. Namba, [0074]-[0087]. Figure 5 illustrates a sound source that is determined to be closer to microphone M2 and located on “hyperbola 1” at position (x, y) based on the volume difference (magnitudes of the first and second sound pick-up signals received at M1 and M2). Namba, [0074], FIG. 5.

In another example, the “sound source position of an input sound is estimated based on volumes of two input sounds” using *inter alia* “the inverse square law” and “the ratio of volumes arriving at the microphones” to determine the sound source is located on a circle closer to microphone M6. Namba, [0078]-[0083], FIG. 7. In yet another example, similar calculations are performed to determine that the sound source is positioned at the intersection of two circles. Namba, [0086]-[0087], FIG. 8.

a. Motivation to Combine

As explained for claims 2 and 14, the techniques implemented by Namba’s distance/direction estimator are particularly useful for determining sound input direction because they *combine* phase *and* volume (magnitude) “estimation methods” to detect sound input direction “even if” the sound input signals (sound pick-up signals) experience “reverberation” or “reflection” caused by the sound pick-up apparatus itself. Namba, [0070]. Thus, to the extent Patent Owner disputes

or the Board finds that Kale does not disclose claims 5 and 17, a POSITA would have been motivated to also detect the voice incoming direction based on magnitudes of the first and second sound pick-up signals to enhance direction detection during “reverberation” or “reflection.” Namba, [0070]; Lipoff, ¶136.

b. Expectation of Success

Kale and Namba use similar elements and techniques to achieve similar results (*see* Overview of Namba, *supra*) and a POSITA would have had a reasonable expectation of success in configuring Kale’s voice direction detector to implement Namba’s magnitude-based voice direction determination process because it would have merely required routine skill in the art to do so. For example, a POSITA would have known to implement the formulas and questions identified in [0074]-[0083] of Namba, taking into account the microphone position of the noise suppression device of Kale, to ascertain voice incoming direction based on magnitude. Lipoff, ¶137.

4. **Claims 6 and [18]: “wherein the adaptive filter performs the noise reduction process [the noise reduction process is performed] to reduce a noise component carried by the first sound pick-up signal using the second sound pick-up signal when the first sound pick-up signal has a greater magnitude than the second sound pick-up signal**

whereas the adaptive filter performs the noise reduction process [the noise reduction process is performed] to reduce a noise component carried by the second sound pick-up signal using the first sound pick-up signal when the second sound pick-up signal has a greater magnitude than the first sound pick-up signal.”

Kale and Namba render obvious claims 6 and 18 because Kale’s beamformer (adaptive noise filter) reduces the noise component of whichever signal contains the more prominent signal (either the first or second sound pick-up signal) using the noise signal (the other of the first or second sound pick-up signal), as explained for Ground 1. Kale, [0034], [0039]; Lipoff, ¶¶138, 147. Modifying Kale to incorporate Namba’s magnitude-based voice direction detection, as described in claim 5, does not alter this.

5. **Claims 8 and [19]: “wherein the voice direction detector detects the voice incoming direction [the voice incoming direction is detected] based on a phase difference between the first and second sound pick-up signals and magnitudes of the first and second sound pick-up signal.”**

Kale and Namba render obvious claims 8 and 19 because the combination teaches a voice direction detector that uses volume/amplitude (magnitude) and

phase difference to determine voice direction, as explained in claims 2, 5, 14, and 17. Namba, [0070]; Lipoff, ¶¶139, 148.

6. **Claims 9 and [20]: “wherein the speech segment determiner detects the speech segment [the speech segment is detected]based on the first sound pick-up signal when the first sound pick-up signal has a more advanced phase than the second sound pick-up signal**

whereas the speech segment determiner detects the speech segment [the speech segment is detected] based on the second sound pick-up signal when the second sound pick-up signal has a more advanced phase than the first sound pick-up signal.”

Kale-Namba renders obvious claims 9 and 20 because Namba discloses a “sound type estimation unit” that detects voice (a speech segment) based on sound pick-up signal phases (delays) calculated by its “distance/direction estimator.”

Namba, [0060], [0071], [0089], [0091]; Lipoff, ¶¶140-143, 149.

Namba discloses a speech segment determiner because it discloses a “sound type estimation unit that *determines whether any sound* originating from the neighborhood of the imaging apparatus such as a voice of an operator of the imaging apparatus or noise resulting from an operation of the operator *is contained.*” Namba, [0060] (emphasis added). Namba further discloses that its sound type estimation unit detects the voice (speech segment) based on the sound pick-up signal phases (delays) calculated by its “distance/direction estimator.”

Namba, [0060] (“The sound type (steady or non-steady, noise or sound) is estimated, for example, from...direction distance information.”).

For example, and as explained for claim 2, *supra*, Namba’s distance/direction estimator measures “the phase of each input sound reaching a microphone,” e.g., “M1 and a microphone M2 ... and a phase difference of the input sounds” and uses the phases and phase difference to calculate “the sound source position of input sound.” Namba, [0071]. The direction information is then fed into a “sound estimator” of the sound type unit to determine if the sound is voice (e.g., the device operator’s voice). Namba, [0089] (“The sound estimator 146 collectively determines whether any neighborhood sound originating from a specific sound source in the neighborhood of the sound processing apparatus 10 *such as a voice of the operator...* is contained in the input sound based on at least one of the ... *positional information of input sound.*” (emphases added)). If a sound input signal is advanced, the estimator may determine that the input contains voice. Namba, [0091] (“If the position of the sound source of input sound is behind an imaging unit^[7] ... and the input sound has sound quality that

⁷ Sound received “behind” the imaging unit (rear microphone) is advanced because it is received closer to another microphone. Namba, [0041], [0077]; Lipoff, ¶142.

matches or resembles that of human voice, the sound estimator 146 may determine that the voice of the operator is [the] predominant[...sound input.”).

Namba’s direction determination improves sound input detection even when sounds reverberate or reflect off the user device. Namba, [0070]. Thus a POSITA would have been motivated and found it obvious to implement Namba’s direction/determination unit functionality to improve Kale’s speech detection. Lipoff, ¶143. A POSITA would have had a reasonable expectation of success in making such a modification because it would have merely required routine skill in the art to do so, as explained for claim 2. Lipoff, ¶143.

VII. Ground 3: Claims 2-4, 7, and 14-16 Are Obvious over Kale and Li-638

A. Li-638 (Ex. 1009)

Li-638, filed Dec. 15, 2010 and published August 2, 2012, is prior art to the ’259 patent under §102(e). Li-638, cover. Li-638 is analogous art to the ’259 patent because it is in the same field of endeavor as the ’259 patent: noise reduction. *Compare* Li-638, [0001], *with* Ex. 1001, 1:17-19; Lipoff, ¶78. Furthermore, Li-638 is directed to the same problem as the ’259 patent: improving noise reduction methods to enhance voice sound quality. *Compare* Li-638 [0010], *with* Ex. 1001, 1:38-42. Lipoff, ¶178.

B. Claims

1. **Claims 2 and [14]: “the voice direction detector detects the voice incoming direction [voice incoming direction is detected] based on a phase difference between the first and second sound pick-up signals.”**

Kale-Li-638 renders claims 2 and 14 obvious because Li-638 teaches “a noise reduction control device” comprising a “filtering control unit” for “calculating incidence angles of []microphone array signals” (target speech and noise signals) based on the relative time delays of [] frequency bins or sub-bands” and “calculating incidence angles of the microphone array signals *based on the relative time delays* of the frequency bins or sub-bands.” Li-638, [0030], [0053], [0061]; Lipoff, ¶150-156, 164.

Like Kale, Li-638’s noise reduction control device includes two microphones (“mic_a and mic_b”) for receiving “signals s_1 and s_2 ... wherein the signal s_1 is treated as desired speech signal and s_2 as reference signal.” Li-638, [0052]. The noise reduction device also includes a filtering control unit (voice direction detector) that determines the “incidence angle” of the “target speech component” (voice incoming direction) by calculating “phase differences between various frequency bins or sub-bands of the microphone array signals and calculating relative time delays of the frequency bins or sub-bands of the

microphone array signals from the phase differences.” Li-638, [0030], [0053], [0061], FIG. 5.

More specifically, the filtering control unit (voice direction detector) includes (1) “a signal delay estimation unit for calculating *phase differences between various frequency bins or sub-bands of the microphone array signals and calculating relative time delays of the frequency bins or sub-bands of the microphone array signals from the phase differences*” and (2) “a signal direction estimation unit for calculating incidence angles of the microphone array signals *based on the relative time delays of the frequency bins or sub-bands*” Li-638, [0030], [0053], [0061], FIG. 5 (emphases added). Li-638 further explains:

Assuming the target speech signal is incident from 0 degree direction, the signal direction estimation unit converts [the] relative time delays of each of [the] frequency bins or sub-bands of the two signals into their incidence angle [such that] target speech components within [an] angle of protection and noise components outside the angle of protection may be distinguished according to their incidence angles. Li-638, [0053].

Thus, Li-638 teaches a voice direction detector that detects the voice incoming direction based on a phase difference between the first and second sound pick-up signals.

a. Motivation to Combine

Li-638 explains that its “technical solution for noise reduction by means of voice incidence angles” (phase difference-based voice direction determination) “protects the target speech while eliminating noises in the target speech” by allowing the adaptive filter to slow down and eventually stop the filtering processes when “all signals are target speech components” Li-638, [0055], [0065]. Such techniques would have been advantageous over prior art systems, which were unable to adequately account for the fact that “signals received by both microphones contain speech components” and were therefore prone to damaging the desired speech signal during the noise reduction process. Li-638, [0004]; Lipoff, ¶154.

Accordingly, a POSITA would have been motivated to modify Kale’s DOA module to implement (or substitute Kale’s DOA module for) Li-638’s signal delay estimation, signal direction estimation, and signal component statistic units (voice direction detector), to improve Kale’s noise reduction process by reducing the likelihood of removing desirable speech from the input signals using Li-638’s voice direction determination techniques. Lipoff, ¶155.

b. Expectation of Success

Kale and Li-638 use similar elements and techniques to achieve similar results (see overview of Li-638, *supra*) and a POSITA would have had a

reasonable expectation of success in configuring Kale's noise suppression device to implement Li-638's voice direction determination process because it would have merely required routine skill in the art to do so. For example, a POSITA would have known to integrate Li-638's voice direction detector such that it receives Kale's first and second sound input signals (speech and noise signals 845, 850) as input into the signal delay estimation, because Li-638's signal delay estimation, like Kale's DOA Module, is configured to receive speech and noise signals from an FFT unit/module. Kale, [0031]; Li-638, [0030]; Lipoff, ¶156.

- 2. Claims 3 and [15]: “the adaptive filter performs the noise reduction process[the noise reduction process is performed] to reduce a noise component carried by the first sound pick-up signal using the second sound pick-up signal when the first sound pick-up signal has a more advanced phase than the second sound pick-up signal**

whereas the adaptive filter performs the noise reduction process [the noise reduction process is performed] to reduce a noise component carried by the second sound pick-up signal using the first sound pick-up signal when the second sound pick-up signal has a more advanced phase than the first sound pick-up signal.”

Kale and Li-638 render obvious claims 3 and 15. Lipoff, ¶¶157-160, 165.

As explained for claim 2, Li-638's voice direction determiner calculates the “time delay” and “incidence angle” of the noises and targeted speech contained in each of the sound pick-up signals s1, s2 so that its adaptive filter H can reduce noise in the signals until they “contain only target speech components.” Li-638,

[0064]. The time delay and incidence angle (phase) are indicative of the amount of noise or speech contained in the signal's sub band, such that when the incidence angle is large (with respect to "target speech that is in 0 degree direction"), the sub band contains more noise. Li-638, [0064] [0069] ("The larger the incidence angle is, the more the speech of this frequency bin or sub-band deviates from the target speech that is in 0 degree direction."). Conversely, "[w]hen the incidence angle of the i^{th} frequency bin or sub-band is in the 0 degree direction" it contains target speech only. Li-638, [0064], [0069]. Thus, a POSITA would have understood that Li-638's adaptive filter reduces the noise component of signals (s1 or S2) having lower incidence angles (advanced phase and less noise) using the noisier signal having the delayed phase (large incidence angle). Li-638, [0064] [0069].

a. Motivation to Combine

As explained for claim 2, Li-638's "technical solution for noise reduction by means of voice incidence angles" (phase difference-based voice direction determination) "protects the target speech while eliminating noises in the target speech" by allowing the adaptive filter to slow down and eventually stop the filtering processes when "all signals are target speech components" Li-638, [0055], [0065]. A POSITA would have been motivated and found it obvious to modify Kale's adaptive filter (as described in [1d]) to implement Li-638's phase-based

noise suppression process to improve the noise reduction as explained in claim 2.

Lipoff, ¶159.

b. Expectation of Success

Kale and Li-638 use similar elements and techniques to achieve similar results (see overview of Li-638, *supra*) and a POSITA would have had a reasonable expectation of success in configuring Kale's noise suppression device to implement Li-638's phase-based adaptive filtering process because it would have merely required routine skill in the art to do so as explained for claim 2.

Lipoff, ¶160. For example, a POSITA would have known to implement the algorithms identified in [0063]-[0064] of Li-638 to control Kale's adaptive filter to eliminate noise from the desired advanced signal (target speech signal) without harming the speech therein as taught by Li-638. Lipoff, ¶160.

3. Claims 4 and [16]: “wherein when the phase difference is within a predetermined range, the adaptive filter outputs either the first or the second sound pick-up signal without performing the noise reduction process.”

Kale and Li-638 render obvious claims 4 and 16 because Li-638 further teaches that its adaptive filter slows down and eventually stops the filtering processes when “all signals are target speech components” having an incidence angle in the range of -45° to 45° . Li-638, [0055], [0061], [0065]; Lipoff, ¶¶161-162, 166.

Li-638 explains that its adaptive filter slows down as the number of target speech signals in a received set of signals increases, and ceases updating entirely—that is, outputs the target signal without performing noise reduction—when “all signals are target speech components, $\alpha=0$.” Li-638, [0055] (“[T]he more the target signal components, which are within the angle of protection, the smaller α is, the slower the updating of the adaptive filter is. When all signals are target speech components, $\alpha=0$, the adaptive filter stops updating of weights of the filter in this speech section, thereby protecting speech in the desired speech signal s_1 from being balanced out, thus effectively protecting target speech from being damaged.”).

Li-638 explains that its filter stops updating (thereby outputting the target signal without performing noise reduction) to avoid damaging the target speech signal, and a POSITA would have been motivated and found it obvious to do the same in the Kale-Li-638 combination. Li-638, [0055]; Lipoff, ¶162.

4. **Claim 7: “The noise reduction apparatus according to claim 5, wherein when a power difference that is a difference between magnitudes of the first and second sound pick-up signals is within a predetermined range, the adaptive filter outputs either the first or the second sound pick-up signal without performing the noise reduction process.”**

Li-638 renders claim 7 obvious because similar to claim 4, Li-638 teaches that its adaptive filter “stops updating,” i.e., stops “suppressing noise” when the “energy” (magnitude) difference between the noise signal s_3 and speech signal s_1

is at a minimum. Li-638, [0045], [0048], [0056]. Lipoff, ¶163. A POSITA would have been motivated and found it obvious to output the target signal without performing noise reduction when the energy is at a minimum (in a predetermined range), as taught by Li-638, to avoid damaging the target speech signal. Li-638, [0055]; Lipoff, ¶163; claim 4.

VIII. Ground 4: Claims 10 and 11 Are Obvious over Kale and Yamada

A. Yamada (Ex. 1010)

Yamada, filed June 16, 2011, and published May 24, 2012, is prior art to the '259 patent under §102(e). Yamada, cover. Yamada is analogous art to the '259 patent because it is in the same field of endeavor as the '259 patent: noise reduction. *Compare* Yamada, [0001], *with* Ex. 1001, 1:17-19; Lipoff, ¶80. Furthermore, Yamada is directed to the same problem as the '259 patent: improving noise reduction methods to enhance voice sound quality. *Compare* Yamada [0011], [0013], *with* Ex. 1001, 1:38-42; Lipoff, ¶80.

B. Claims

- 1. Claim 10: “The noise reduction apparatus according to claim 1, wherein signals are supplied to the voice direction detector as the first and second sound pick-up signals at a sampling frequency of 24 KHz or higher and signals are supplied to the adaptive filter as the first and second sound pick-up signals at a sampling frequency of 12 KHz or lower.”**

Kale discloses supplying the sound pick-up signals to its beamformer (adaptive filter) at a sampling frequency of 12 KHz or lower because Kale discloses supplying 8KHz signals: Kale, [0029] (“According to the exemplary embodiments, the FFTs 805, 810 may compute a 128, 256, and/or 512 point FFT of a 8 kHz signal”). Lipoff, ¶167. Kale further explains that the sampling rate “may be changed” depending on the “resolution desired” of the output signals. Kale, [0029].

Yamada teaches a signal processing method wherein “sound picked up by a microphone” is supplied to a “sound source direction estimation section” “at a sampling frequency of 48 kHz.” Yamada, [0067]-[0069]; Lipoff, ¶169.

a. Motivation to Combine

A POSITA would have been motivated and found it obvious to supply the sound pick-up signal of Kale to the DOA module at a sampling rate of 24 KHz or more, including at 48KHz, as taught by Yamada, to improve the accuracy of Kale’s voice direction determination. Lipoff, ¶170. Indeed, a POSITA would have

understood that increasing the sampling rate would have been particularly useful for voice direction determination because lower rates, such as 8 KHz, while suitable for noise cancellation and suppression, would have produced low-resolution signals that would have made it more difficult to pinpoint the voice direction. Lipoff, ¶170. Furthermore, it was known in the art to adapt sampling rate depending on the desired accuracy of the signal. Lipoff, ¶170 (citing Pedersen, [0023] (“In general, the sampling frequency is adapted to the application (available bandwidth, power consumption, frequency content of input signal, necessary accuracy, etc.). In an embodiment, the sampling frequency f_s is in the range from 8 kHz to 40 kHz, e.g. from 12 kHz to 24 kHz, e.g. around or equal to 16 kHz or 20 kHz.”)).

b. Expectation of Success

A POSITA would have had a reasonable expectation of success in configuring Kale’s noise cancellation device to supply sound input signals to the DOA module at 24 KHz or more because it would have merely required routine skill in the art to do so. Lipoff, ¶171.

A POSITA would have known to implement e.g., a sample frequency converter or down sampling unit to continue to supply the sound input signal to Kale’s adaptive filter as was known in the art. Lipoff, ¶172 (citing Tateyama, [0009]-[0010] (describing a signal processing method wherein an input signal is

down sampled to a lower sampling rate before being provided to an adaptive filter). Maintaining the lower sampling frequency to the adaptive filter would have reduced power consumption on the adaptive filter by reducing the data input.

Lipoff, ¶172 (citing same).

2. **Claim 11: “The noise reduction apparatus according to claim 1, wherein the speech segment determiner outputs more accurate speech segment information to the voice direction detector than speech segment information to the adaptive filter.”**

Kale and Yamada render claim 11 obvious because, as explained for claim 10, increasing the sampling frequency to 48 KHz, as taught by Yamada, increases the accuracy of the outputted signal relative to the 8 KHz sampling frequency.

Lipoff, ¶173.

IX. Ground 5: Claims 1-3 and 12-15 Are Obvious over Li-730

A. Li-730 (Ex. 1005)

Li-730, filed July 28, 2009, and published February 3, 2011, qualifies as prior art to the '259 patent under §§102(a), (e). Li-730, cover. Li-730 discloses an “audio processing apparatus” for “suppressing stationary and non-stationary noises.” Li-730, [0019]. The apparatus includes two microphones (a main and a reference microphone), a voice active detector, a beamformer, and a noise suppressor. Li-730, [0019], FIG. 2A.

As illustrated in Figure 2A below, the main and reference microphones receive a “mixture of sounds”—from a “source” (e.g. a “speaker’s mouth”) and a “non-source” (e.g. “noises”) and “convert the mixture of the sounds and the noises [] into a main input M1 and a reference input M2.” Li-730, [0020]. The converted inputs (M1, M2) are transformed into frequency domain representations (S1, S2), calibrated (into signals C1 and C2), and input into the “voice active detector” (VAD). Li-730, [0021]-[0022], [0025], FIG. 2A. The VAD uses the calibrated signals “to generate a voice active signal V1.” Li-730, [0020], [0025], FIG. 2A.

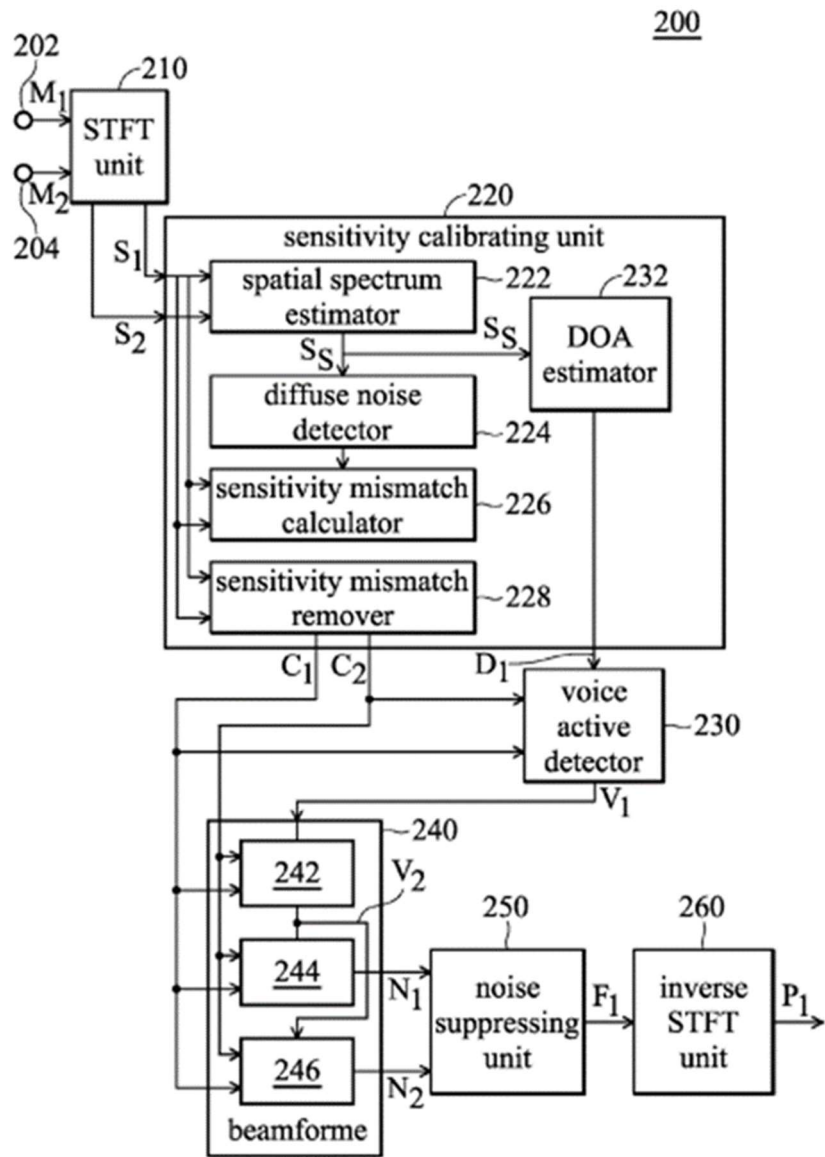


FIG. 2A

Li-730, FIG. 2A.

Li-730's beamformer uses the voice active signal V1 to generate and output "steering vector signal V2" which "indicates [the] directional vector" of the source signal (voice signal) when voice is detected—that is, when the "voice active

signal” is “turned on.” Li-730, [0026], [0031], claim 13, FIGS. 2A, 3C. The beamformer also generates “main” and “reference” channels N1 and N2 (“corresponding to the sounds received from the source” and “the noises received from non-source,” respectively) and outputs the signals to the noise suppressing unit. Li-730, [0026]-[0027], FIG. 2A, 3D. Finally, the noise suppressing unit “suppress[es] stationary and non-stationary noises in the main channel N1 and the reference channel N2 according to the voice active signal V1, and integrate[s] the main channel N1 and the reference channel N2 into a final signal F1.” Li-730, [0027], FIGS. 2A, 3A. Figure 3A below illustrates Li-730’s noise suppression method.

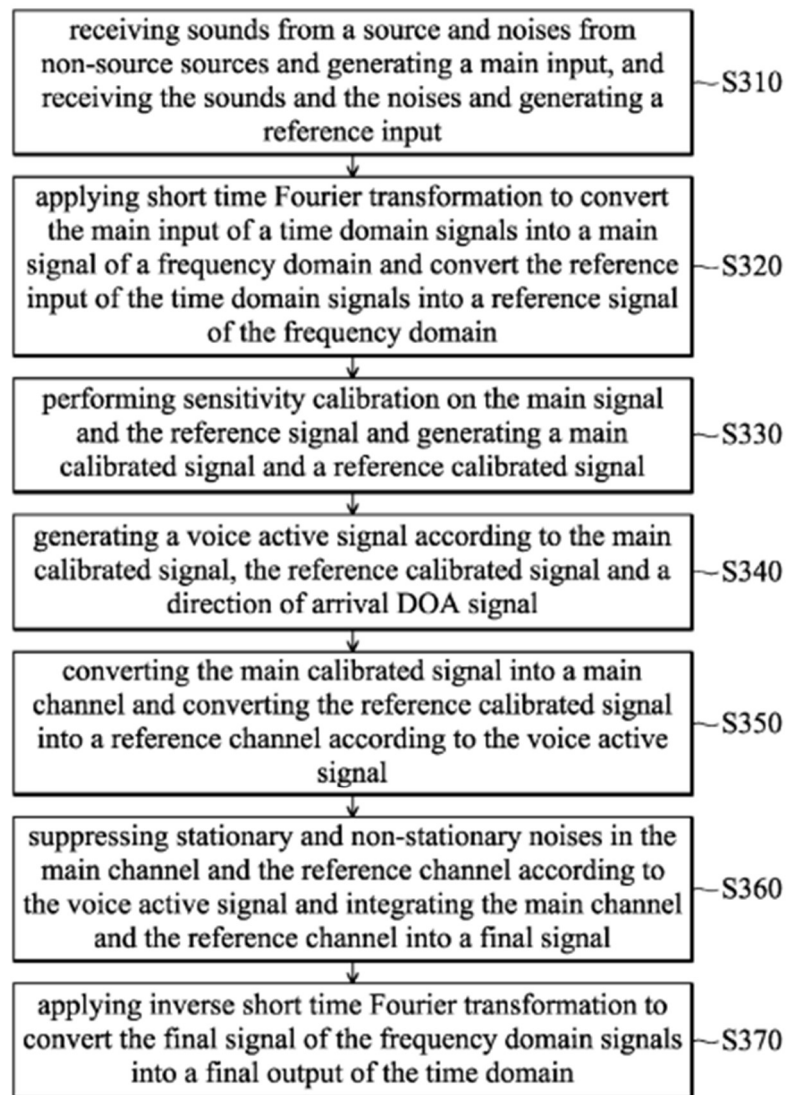


FIG. 3A

Li-730, FIG. 3A.

Li-730 is analogous art to the '259 patent because it is in the same field of endeavor as the '259 patent: noise reduction. *Compare* Li-730, [0002], with Ex. 1001, 1:17-19; Lipoff, ¶69. Furthermore, Li-730 is directed to the same problem as the '259 patent: improving noise reduction methods to enhance voice

sound quality. *Compare* Li-730, [0006] [0028], *with* Ex. 1001, 3:52-59; Lipoff, ¶69.

B. Independent Claims

Li-730 discloses or suggests each element of the independent claims as set forth below (Lipoff, ¶ 69):

1. Claim 1

a. [1preamble]: “A noise reduction apparatus comprising:”

Li-730 discloses [1preamble] because it discloses an “audio processing apparatus” for “suppressing stationary and non-stationary noises.” Li-730, [0019], [0028], FIGS. 2A-2B, 3A. Figure 2A below illustrates the “audio processing apparatus” including a “noise suppressing unit” 250. Li-730, [0019], FIG. 2A; Lipoff, ¶¶174-175.

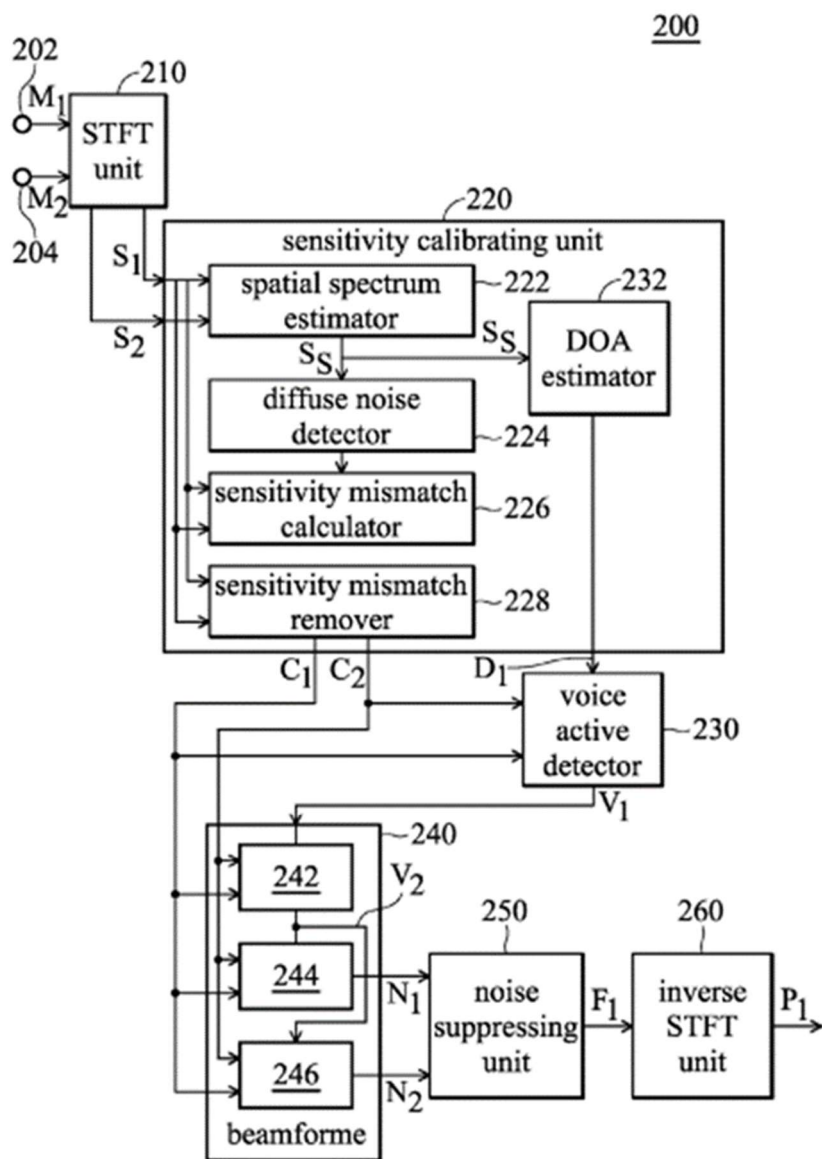


FIG. 2A

Li-730, FIG. 2A.

Figure 2B below illustrates a “cell phone” implementing the disclosed audio processing apparatus. Li-730, [0020].

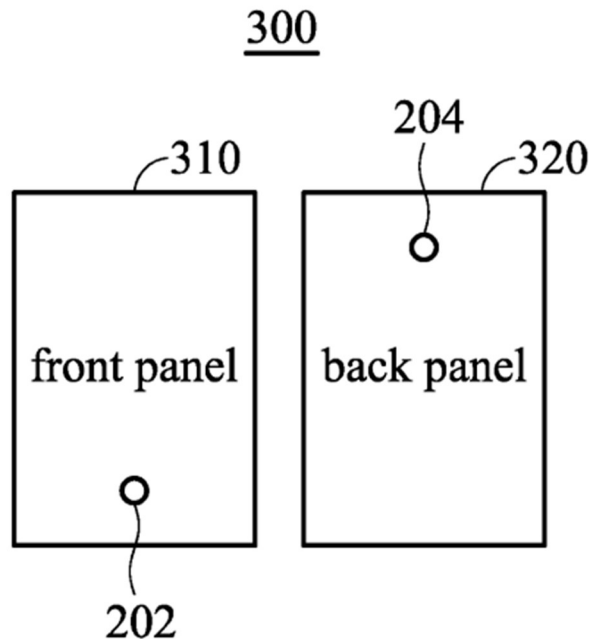


FIG. 2B

Li-730, FIG. 2B.

- b. [1a]: “a speech segment determiner configured to determine whether or not a sound picked up by at least either a first microphone or a second microphone is a speech segment and”

Li-730 discloses [1a] because its audio processing apparatus includes a “voice active detector (VAD)” (speech segment determiner) “for generating a voice active signal” from calibrated “main” and “reference” sound inputs (M1, M2) picked up by “main” and “reference” microphones 202 and 204, respectively.

Li-730, [0007], [0019], [0020], [0025], FIGS. 2A, 3C; Lipoff, ¶¶176-183.

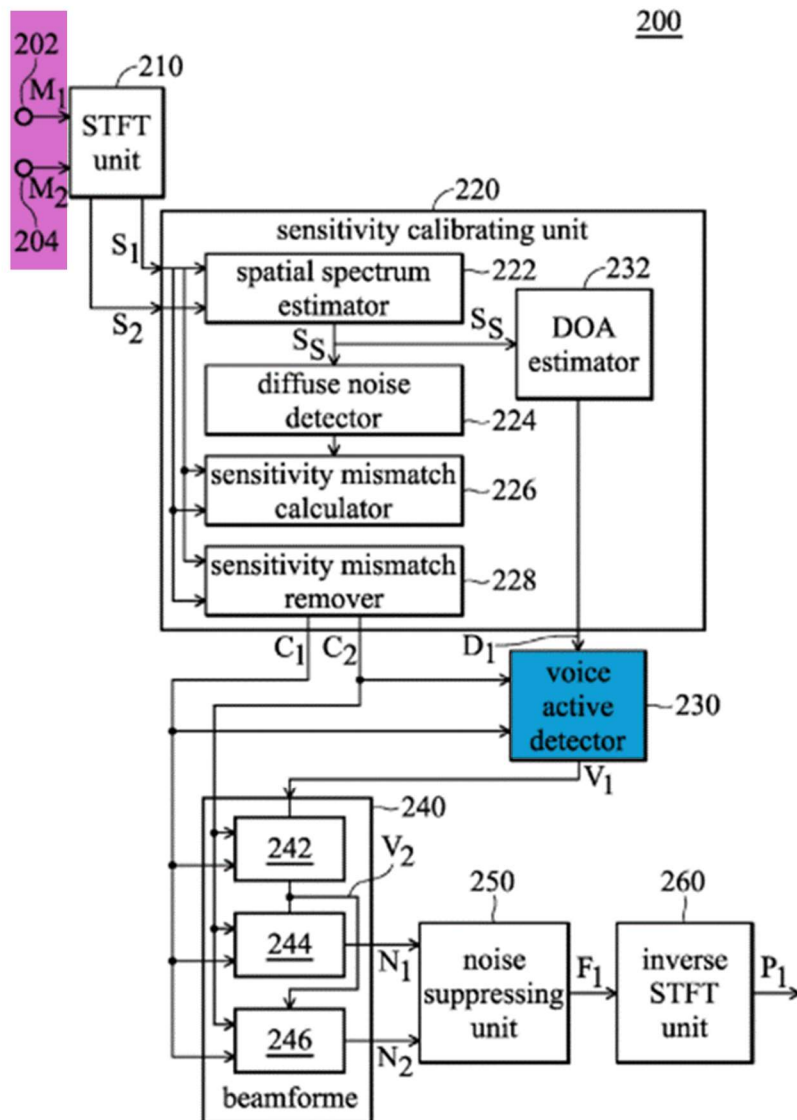


FIG. 2A

Li-730, FIG. 2A (annotated).

Li-730 discloses at least either a first or second microphone for picking up sound as recited in [1a].

As illustrated in Figure 2B below, Li-730's apparatus includes a "main microphone" 202 and a "reference microphone" 204 (first and second

microphones) which may be disposed along front and back panels of, e.g., a cell phone. Li-730, [0020], FIGS. 2A, 2B.

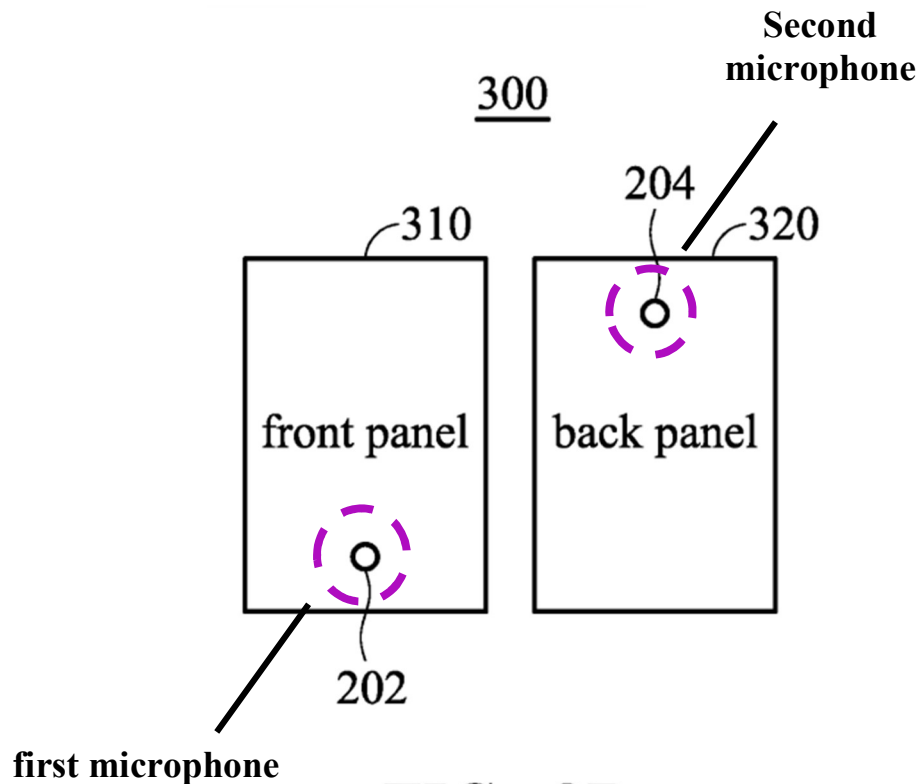
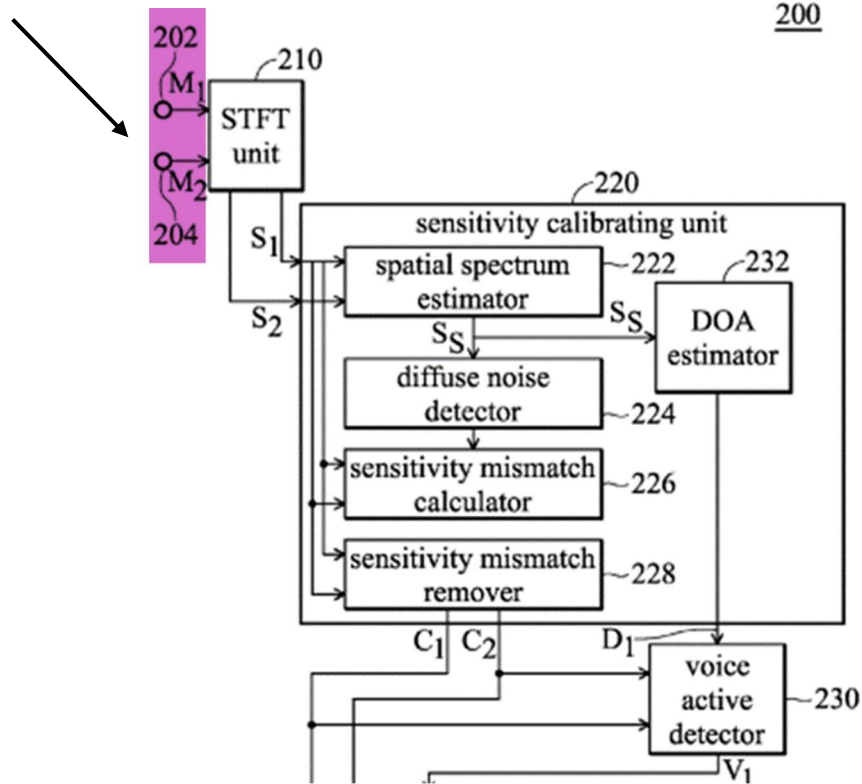


FIG. 2B

Li-730, FIG. 2B (annotated). “The main microphone 202 and the reference microphone 204 are all used to receive sounds”—that is, a “mixture of sounds”—from a “source” (e.g., a “speaker’s mouth”) and a “non-source” (e.g., “noises”). Li-730, [0020]. These sounds are converted “into a main input M1 and a reference input M2 as shown in FIG. 2.” Li-730, [0020], FIG. 2A. Thus, Li-730 discloses sound picked up by at least either a first microphone or a second microphone.

Sound picked up by first
and second microphones 202
and 204



Li-730, FIG. 2A (annotated, excerpted).

Li-730 discloses a VAD (speech determiner) configured to determine whether or not the sound picked up by the main or reference microphone is a speech segment based on a power ratio associated with the sound.

Li-730 explains that the main and reference sound inputs M1, M2 (picked up sounds) are transformed into frequency domain representations (S1, S2), calibrated into signals C1 and C2, and input into the VAD, which uses the calibrated signals

“to generate a voice active signal V1” (a determination that the sound is a speech segment) Li-730, [0025].

More specifically, Li-730 explains that its short-time Fourier transformation (STFT) unit “converts the main input M1 and the reference input M2 of the time domain signals into a main signal S1 and a reference signal S2 of the frequency domain” and outputs the analog signals to “sensitivity calibrating unit 220.” Li-730, [0021]. Once received, the calibrating unit processes the signals (e.g., to generate a “spatial spectrum SS” depicting “the functional relationship between the power distribution and the angles of incident of the main signal and reference signal” and “determine “whether there is a dominant peak in the spatial spectrum SS”) and outputs main and reference calibrated signals (C1 and C2 corresponding M1 and M2) to the VAD. Li-730, [0023]-[0025], FIG. 2B.

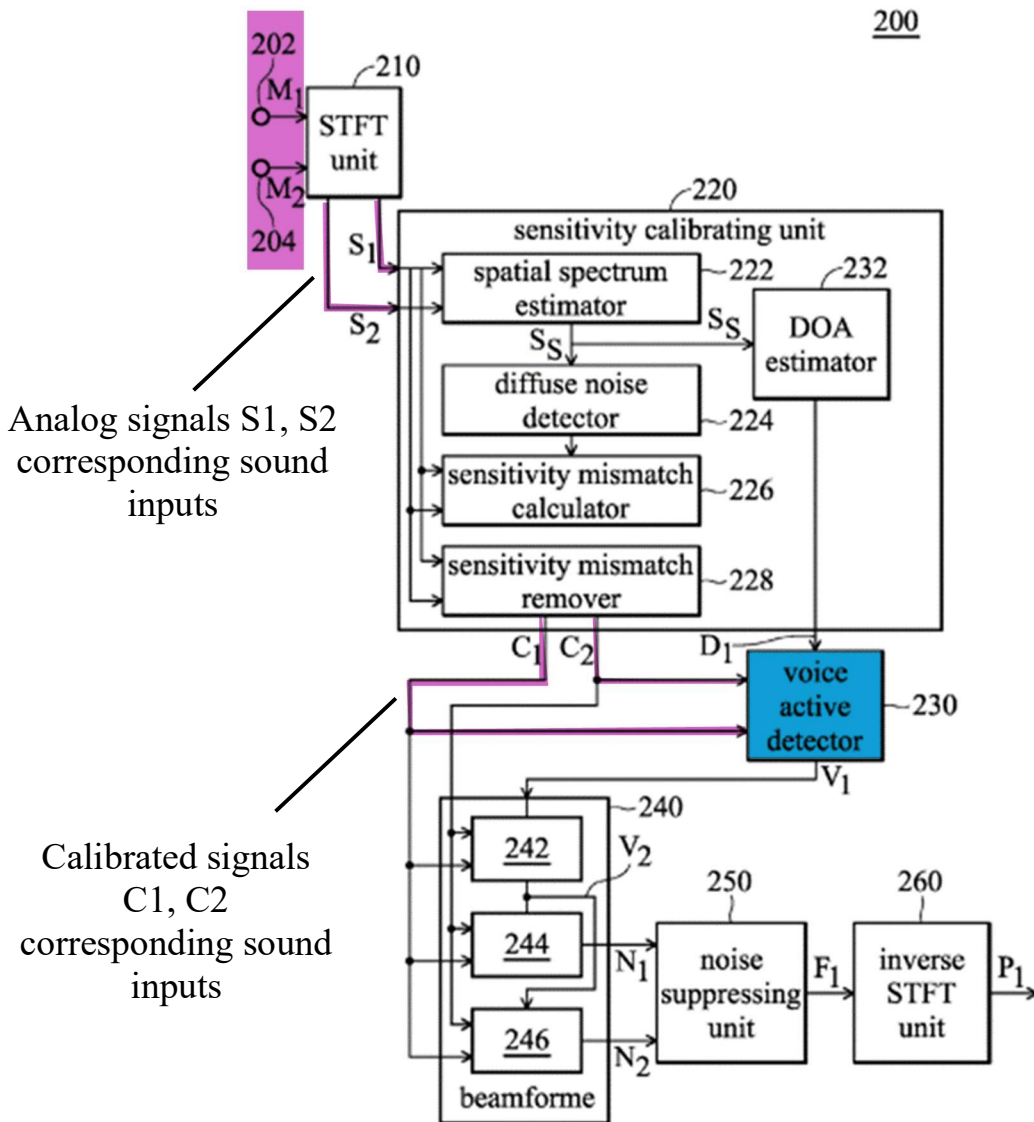


FIG. 2A

Li-730, FIG. 2A (annotated).

The VAD then “compares a power ratio between the main calibrated signal C1 and the reference calibrated signal C2 with a predetermined threshold” to determine whether the power ratio exceeds the predetermined threshold for a particular frequency bin. Li-730, [0025]. The results of this comparison indicate

whether a voice (speech segment)⁸ is present. If “the power ratio in one bin is smaller than the pre-defined threshold, the signals in that bin may be regarded as noises and may be eliminated.” Li-730, [0025]. Conversely, “when the power ratio in one bin is larger than the pre-defined threshold, the signals in that bin may be regarded as *desired signals*,” i.e., voice signals, and “may be preserved.” Li-730, [0025], [0030], FIG. 3C. As illustrated in step S342 of Figure 3C, this determination also causes a “voice active signal” V1 to turn on. Li-730, [0030], FIG. 3.

⁸ The '259 patent indicates that a “speech segment” is a voice input. Ex. 1001, 4:60-66 (“The main microphone 11 and the sub-microphone 12 pick up a sound including a voice component (speech segment).”).

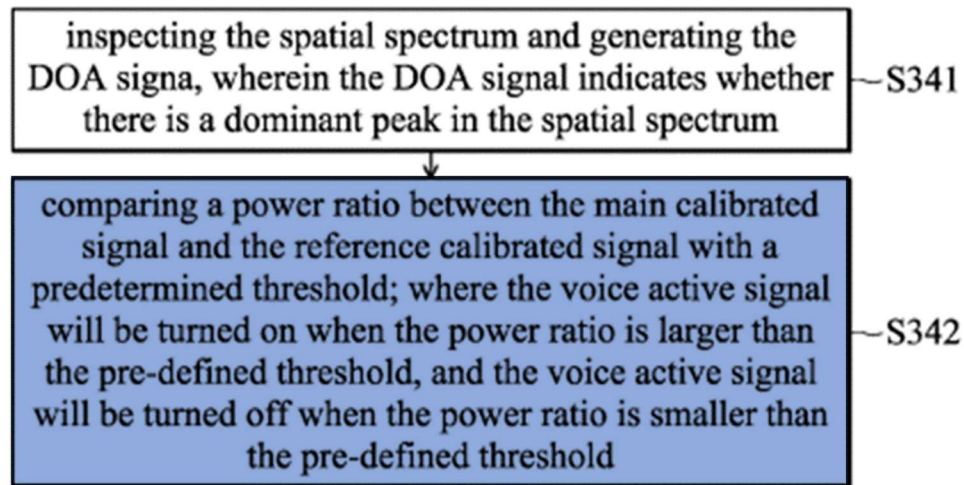


FIG. 3C

Li-730, FIG. 3C (annotated).

Li-730's speech segment identifying process is similar to the technique used in the '259 patent which also compares signal-to-noise ratios (power ratios) to detect a speech segment in certain embodiments. Ex. 1001, 4:59-67 ("[T]echnique I focuses on frequency spectra of a vowel sound that is a main component of a voice sound, to detect a speech segment. In detail, in the speech segment determination technique I, a signal-to-noise ratio is obtained between a peak level of a vowel-sound frequency component and a noise level appropriately set in each frequency band and it is determined whether the obtained signal-to-noise ratio is a specific ratio for a specific number of peaks, thereby detecting a speech segment."); *see also* Ex. 1001, 12:12-20 ("[T]he speech determination

unit 38 []determines whether a per-frame input signal of a target frame is a speech segment, based on the energy ratio.”); Lipoff, ¶182.

Accordingly, Li-730 discloses a speech segment determiner as recited in [1a]. Lipoff, ¶183.

c. [1b]: “to output speech segment information when it is determined that the sound picked up by the first or the second microphone is the speech segment;”

Li-730 discloses [1b] because as explained for [1a], the VAD (speech segment determiner) outputs a “voice active signal V1” (speech segment information) after determining that at least one of the processed calibrated inputs includes voice (a speech segment). Li-730, [0007], [0019], [0020], [0025], [0030]. FIGS. 2A, 3C; Lipoff, ¶184. The voice active signal contains information⁹ about the speech segment that may be used to ascertain its direction and inform noise suppression, as explained in [1c]-[1d], *infra*. Li-730, [0026]-[0028], [0031].

⁹ The ’259 patent does not expressly define “speech segment information” but explains that it “indicat[es] that the sound picked up by the first or the second microphone is the speech segment.” Ex. 1001, Abstract. Patent Owner also contends in its complaint that speech segment information broadly relates to “details/parameters of the speech segment.” Ex. 1014, 9. Li-730’s voice active signal V1 is consistent with these descriptions.

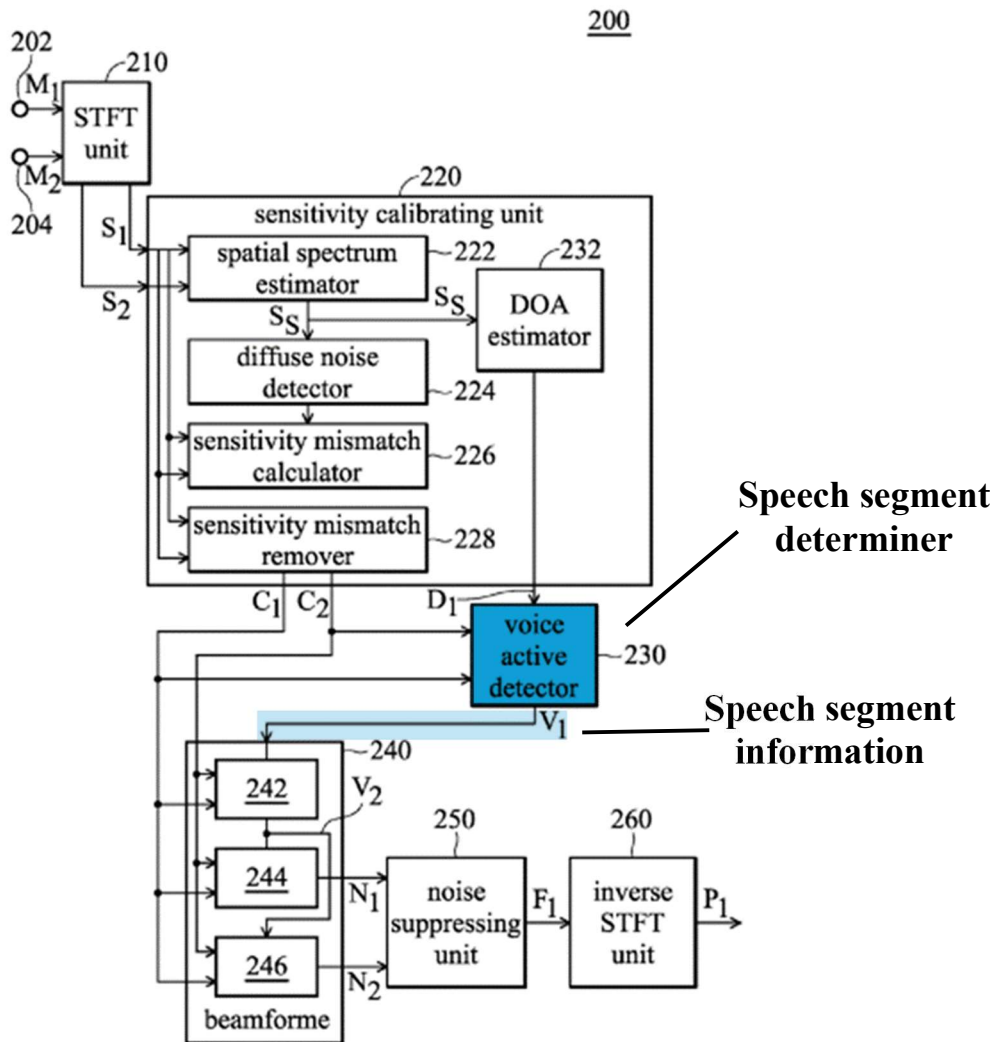


FIG. 2A

Li-730, FIG. 2A (annotated).

- d. [1c]: “a voice direction detector configured, when receiving the speech segment information, to detect a voice incoming direction indicating from which direction a voice sound travels,”

Li-730 discloses [1c] because its audio processing apparatus further includes a “beamformer” (voice direction detector) configured to determine a “directional

vector” for “each frequency bin according to the voice active signal V1” and apply the directional vector to the voice sound (detect a voice incoming direction indicating from which direction a voice sound travels). Li-730, [0026], FIG. 3D; Lipoff, ¶¶185-188.

First, Li-730 discloses a beamformer (voice direction detector) configured to receive speech segment information V1, as illustrated in annotated Figure 2A below. Li-730, [025]-[0026], FIG. 2A; Lipoff, ¶186.

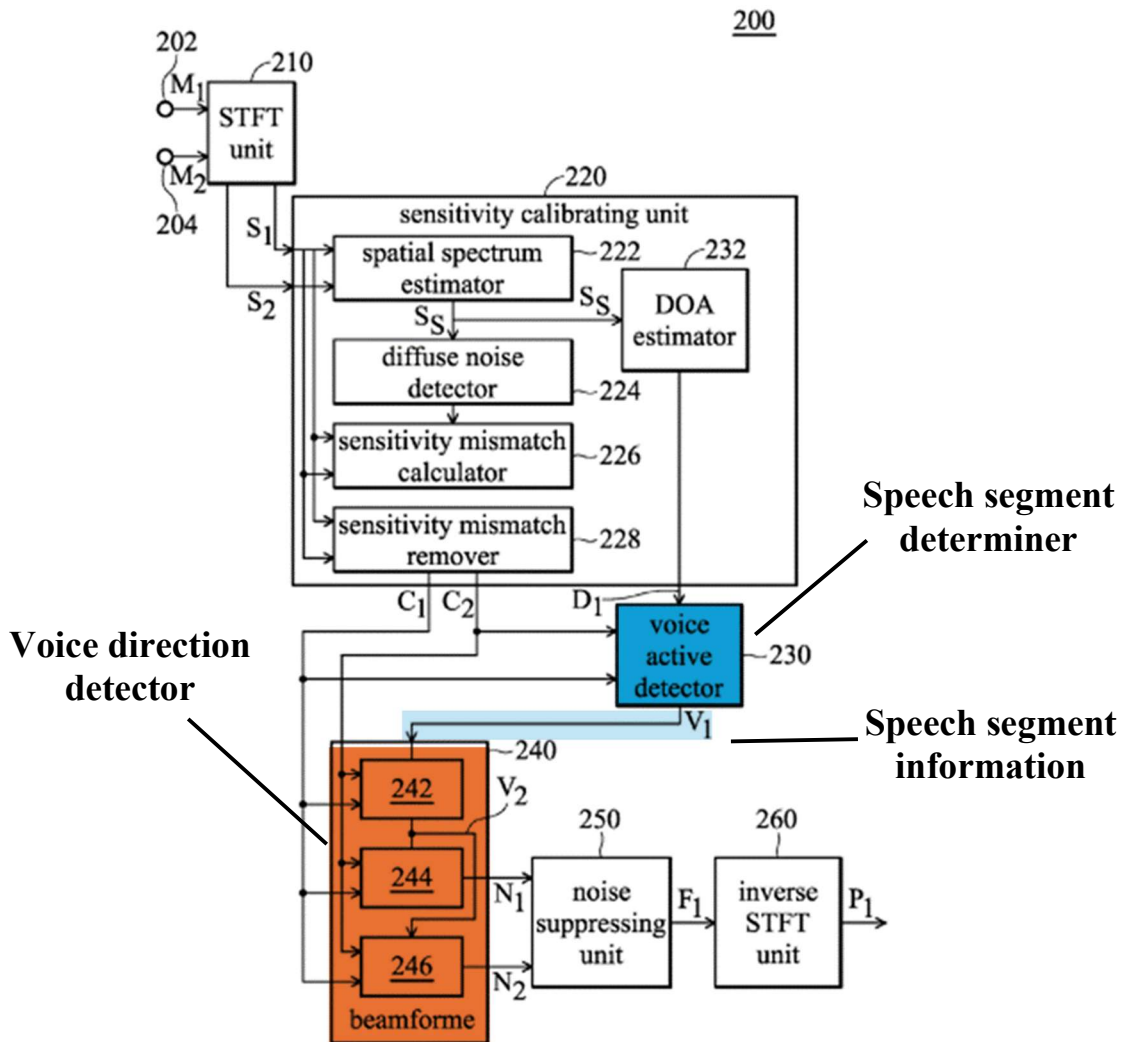


FIG. 2A

Li-730, FIG. 2A (annotated).

Second, Li-730 discloses that the beamformer uses the speech segment information to generate a steering vector signal V_2 , which is in turn used to determine the direction from which the calibrated signal corresponding to voice sound M_1 travels. Li-730, [0026]; Lipoff, ¶187-188.

More specifically, Li-730 explains that the beamformer includes an “array manifold matrix identification unit 242” that implements, e.g., a “projection approximation subspace tracking” (“PAST”) algorithm to “generate a steering vector signal V2 *according to the voice active signal V1*.” Li-730, [0026] (emphasis added), claim 13. This “steering vector signal” V2 “indicates the directional vector” for “each frequency bin according to the voice active signal V1”—in other words, the steering vector determines the direction from which the sounds corresponding to a particular frequency bin travels. Li-730, [0026], [0031].

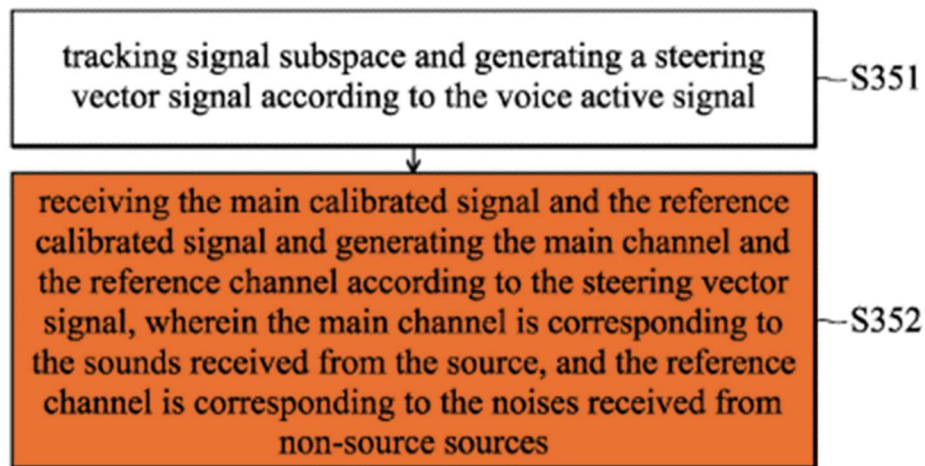


FIG. 3D

Li-730, FIG. 3D (annotated).

- e. **[1c.i]: “based on a first sound pick-up signal obtained based on a sound picked up by the first microphone and a second sound pick-up signal obtained based on a sound picked up by the second microphone”**

Li-730 discloses [1c.i] because, as explained for [1b] and [1c], the voice incoming direction calculation is based on speech segment information V1, which is based on calibrated signals C1, C2 (first and second sound pick-up signals) corresponding to first and second sounds picked up by the “main” (first) and “reference” (second) microphones. Li-730, [0007], [0026], [0031], FIGS. 2A, 3D; Lipoff, ¶189.

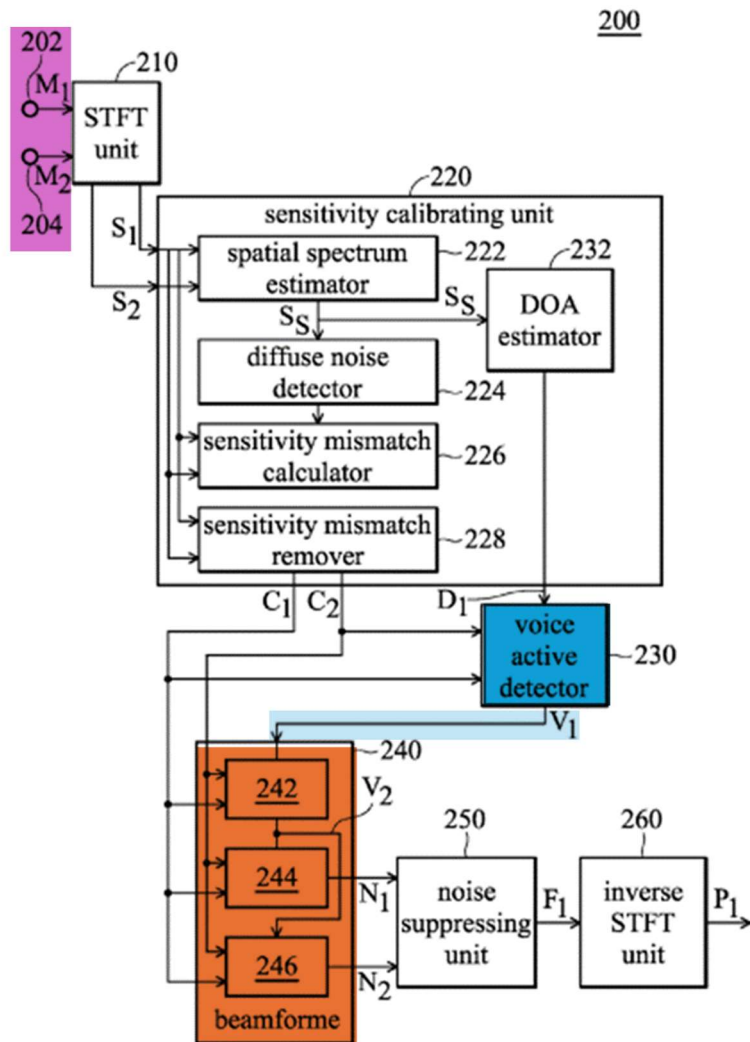


FIG. 2A

Li-730, FIG. 2A (annotated).

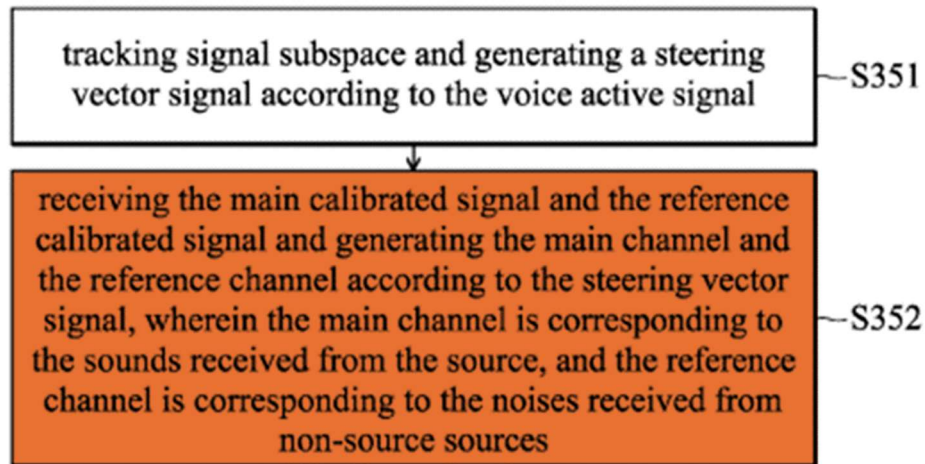


FIG. 3D

Li-730, FIG. 3D (annotated).

- f. **[1c.ii]: “and to output voice incoming-direction information when the voice incoming direction is detected; and”**

Li-730 discloses [1c.ii] because the beamformer (voice direction detector) outputs main and reference channels (voice incoming direction information) based on “steering vector signal V2” which “indicates [the] directional vector” of the source signal (voice signal). Li-730, [0026], [0031], claim 13, FIGS. 2A, 3C (emphasis added); Lipoff, ¶¶190-191.

As illustrated in Figure 2A, the steering vector signal V2 is output to a “main channel calibrator” 244 and “reference channel calibrator 246” where it is then used to generate main channel and reference channel signals N1 and N2. Li-730,

[0026], [0031], FIG. 2A. More specifically, the beamformer applies the steering vector (and a “minimal variance distortion response (MVDR) algorithm”) to calibrated voice signal C1 (as described in [1a]) to “generate” a “main channel [signal] N1” “*corresponding to the sounds received from the source*” (voice). Li-730, [0026], FIGS. 2A, 3D (emphasis added); *see also* [0020] (explaining that the “source” originates from “a speaker’s mouth”).

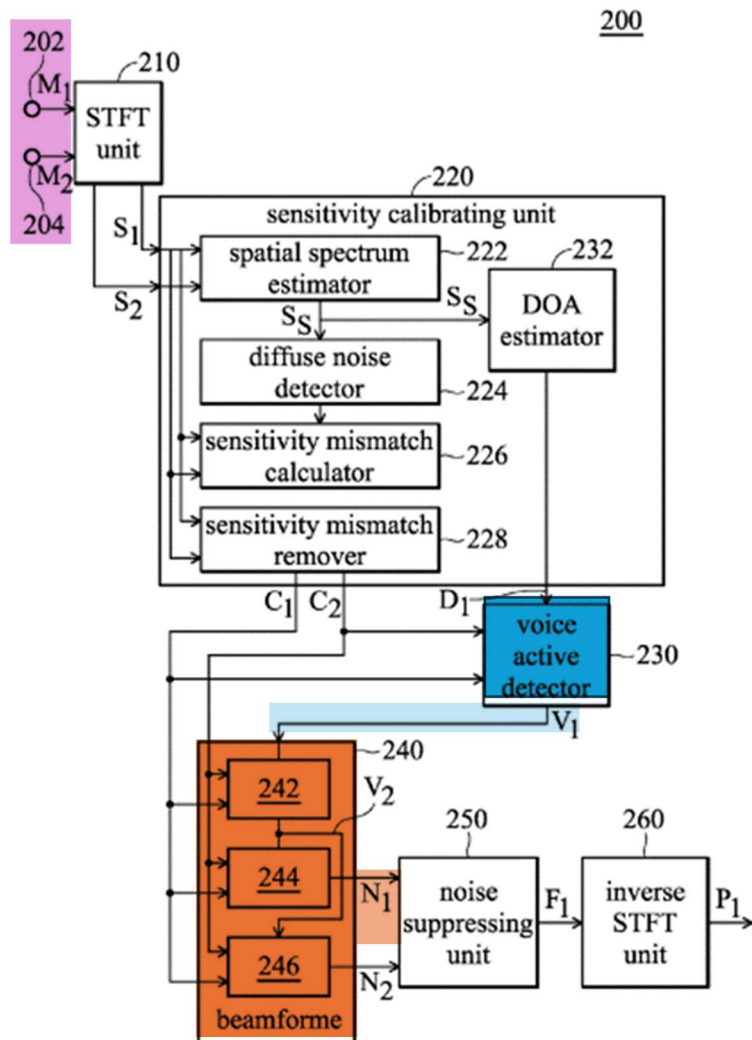


FIG. 2A

Li-730, FIG. 2A (annotated).

- g. [1d]: “an adaptive filter configured to perform a noise reduction process using the first and second sound pick-up signals based on the speech segment information and the voice incoming-direction information.”

Li-730 discloses [1d] because its apparatus includes a “noise suppression unit 250” (an adaptive filter such as a “Weiner post filter”) configured to “suppress

stationary and non-stationary noises” (perform a noise reduction process) using “the main calibrated signal C1 and the reference calibrated signal C2” (the first and second sound pick-up signals), “according to the voice active signal V1” (and based on the speech segment information), and the main and reference channels N1 N2 generated using the steering vector V2 (based on the voice incoming direction information). Li-730, [0027], FIG. 2A; Lipoff, ¶¶192-200.

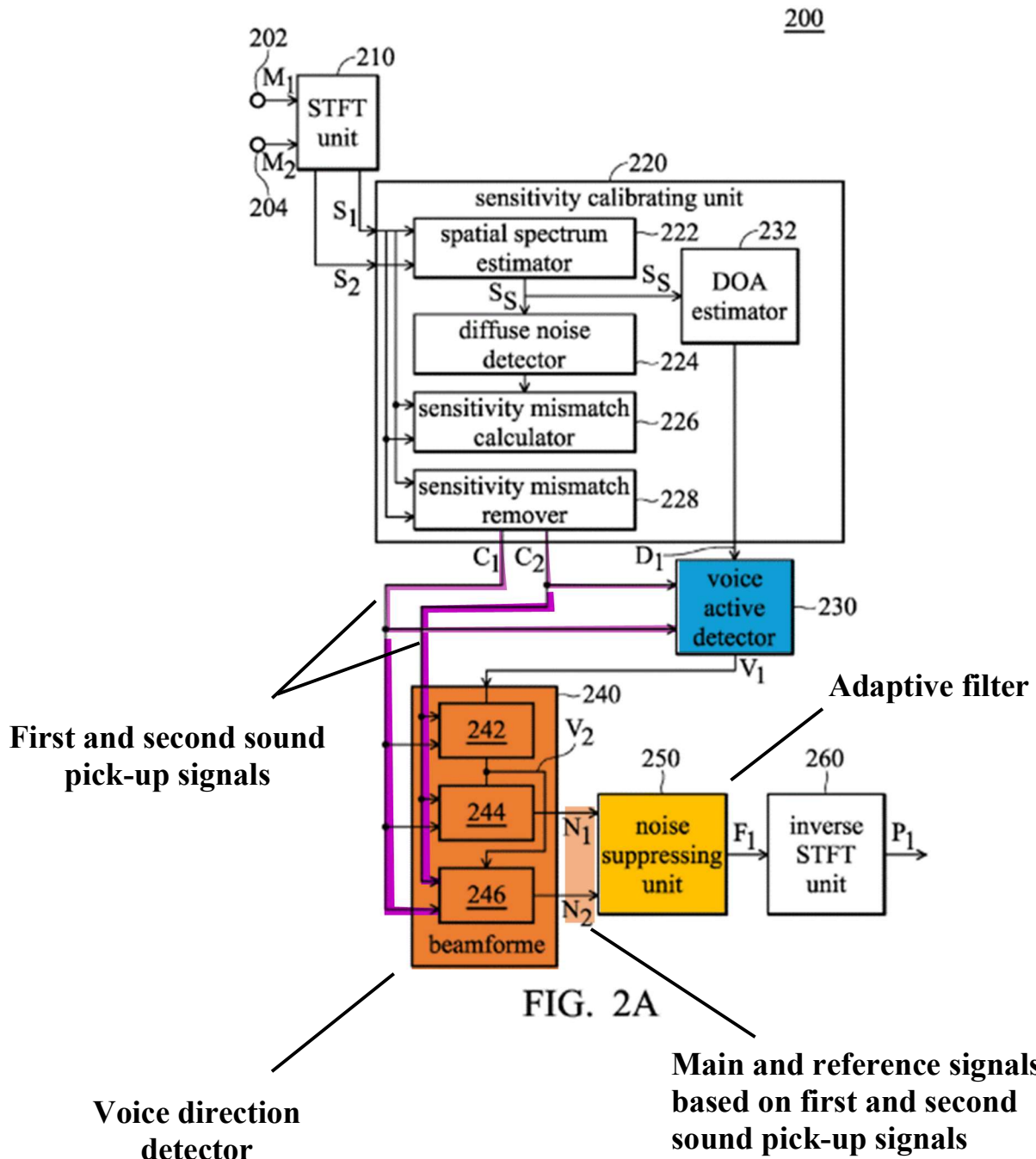
Li-730’s adaptive filter performs noise reduction using the first and second sound pick-up signals.

As discussed above in element [1c.i], Li-730’s signals C1 and C2 correspond to the claimed first and second sound pick-up signals, respectively.

The noise suppression unit 250 uses those signals to reduce noise after some preprocessing. Li-730, [0027]-[0028]; Lipoff, ¶194. As explained in [1c.ii], the beamformer (voice direction detector) applies a “steering vector” and a “minimal variance distortion response (MVDR) algorithm” to “the main calibrated signal C1 and the reference calibrated signal C2” (the first and second sound pick-up signals) to “generate the main channel N1” and “the reference channel N2.” Li-730, [0026]. Main channel N1 “correspond[s] to the sounds received from the source,” i.e., the “speaker’s voice” and reference signal N2 “corresponding to the noises received from non-source sources.” Li-730, [0026], FIGS. 2A, 3D; *see also* [0020] (explaining that the “source” originates from “a speaker’s mouth”). Both

N1 and N2 are output to the noise suppression unit (adaptive filter) which “further suppress[es] stationary and non-stationary noises in the main channel N1 and the reference channel N2 according to the voice active signal V1, and integrate[s] the main channel N1 and the reference channel N2 into a final signal F1.” Li-730, [0027]-[0028], [0031], FIG. 3A.

Thus, as illustrated in annotated Figure 2A below, Li-730’s adaptive filter performs noise reduction using the first and second sound pick-up signals (C1, C2) because they are used to generate the main and reference channels N1 and N2 and “[t]he noise suppressing unit 250 is used to further suppress stationary and non-stationary noises in the main channel N1 and the reference channel N2.” Li-730, [0027], FIG. 2A, 3A; Lipoff, ¶195.



Li-730, FIG. 2A (annotated).

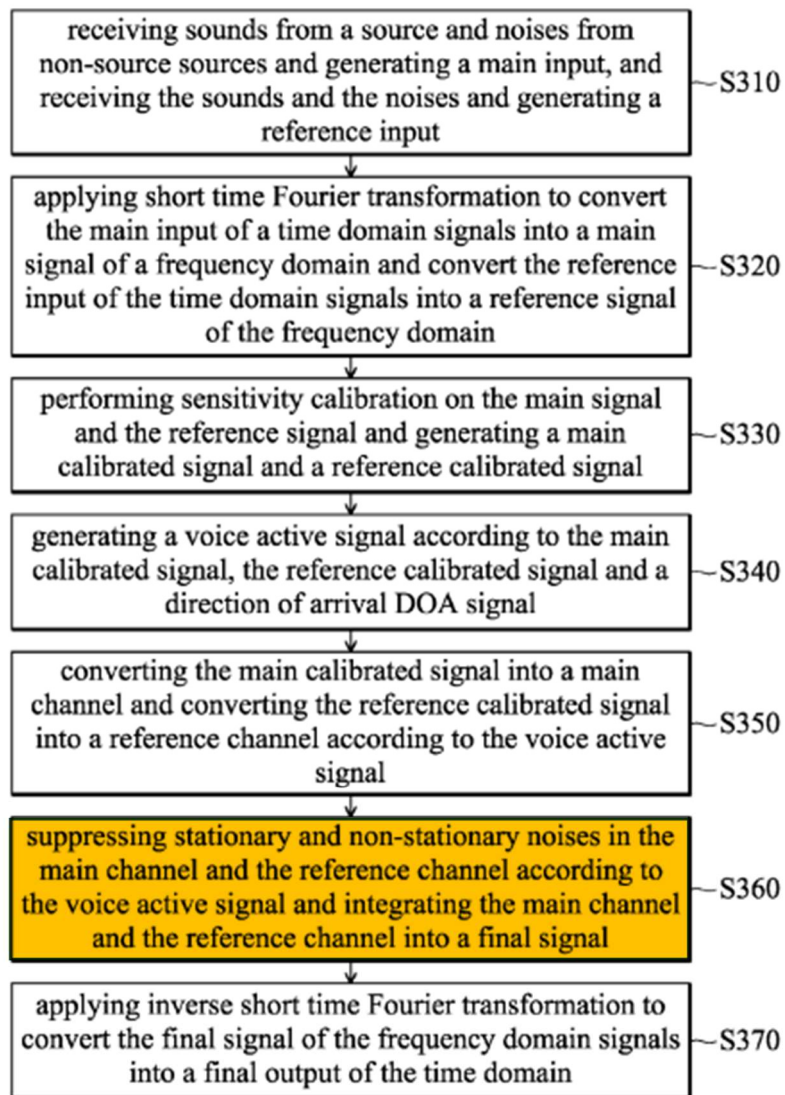


FIG. 3A

Li-730, FIG. 3A (annotated).

Noise suppression is also based on first and second pick-up signals C1 and C2 because they are used to generate the speech segment information V1 (as explained in [1a]) on which noise suppression is based, as explained below. Li-730, [0026]-[0027], FIG. 2A; Lipoff, ¶196.

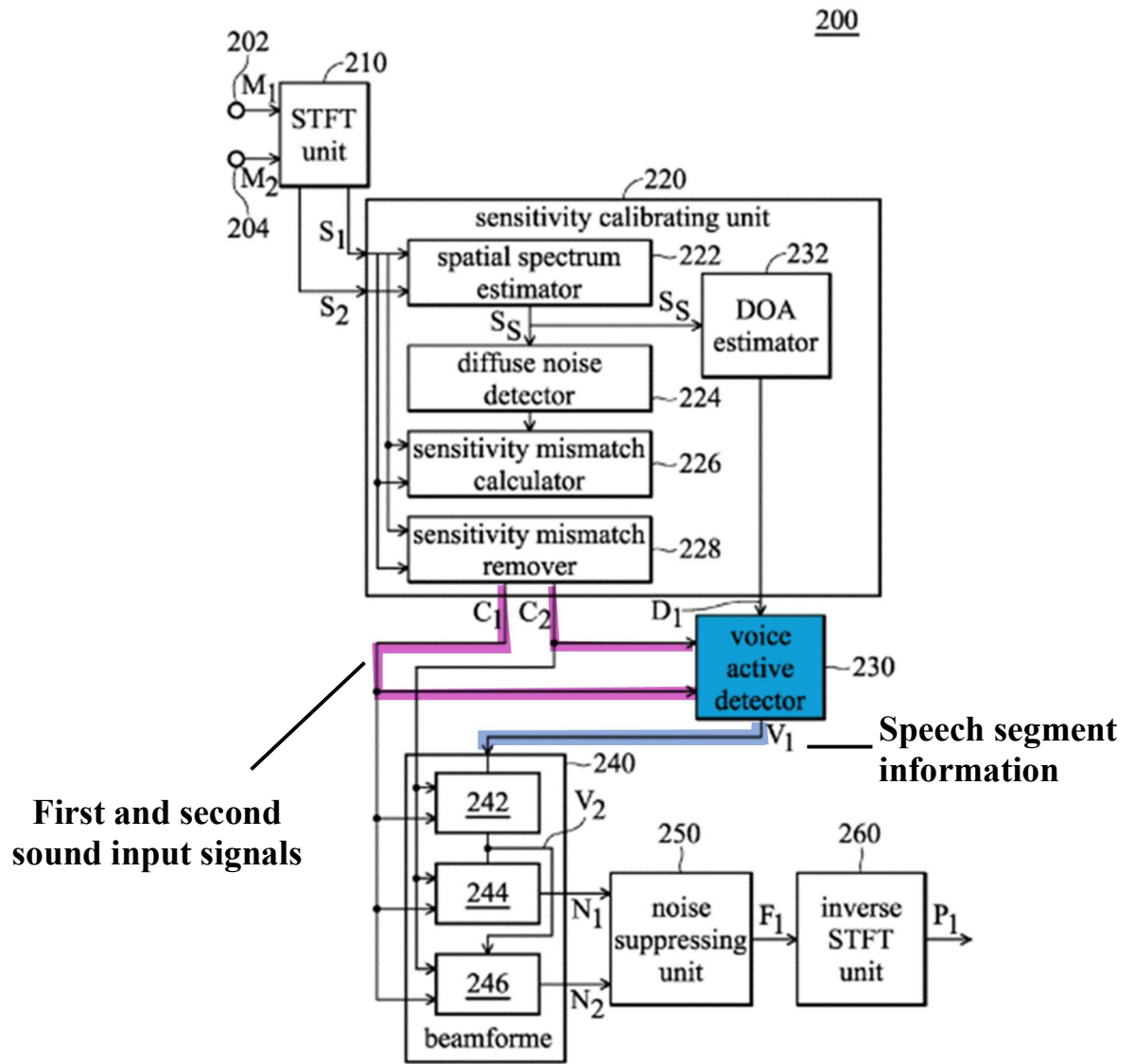


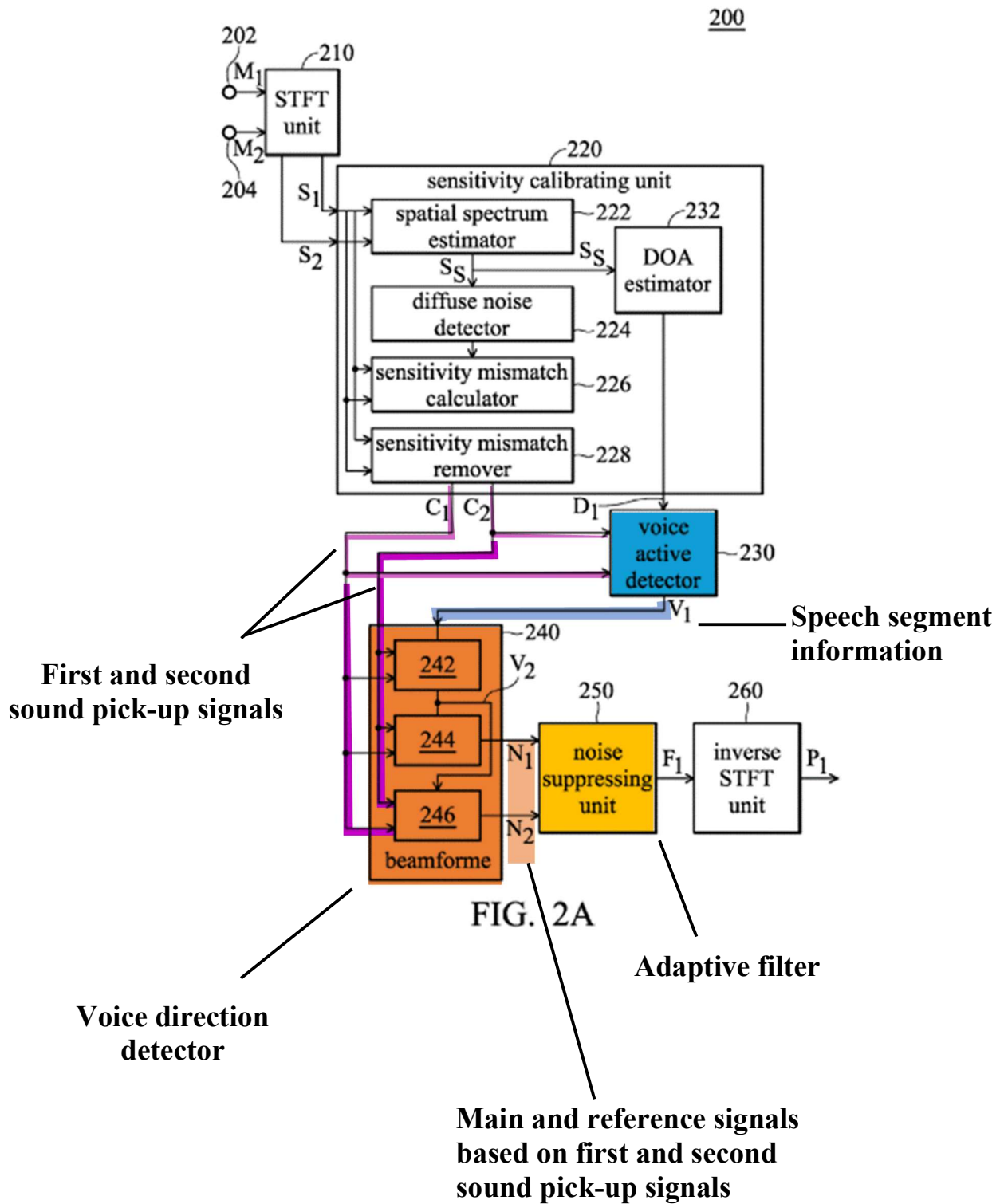
FIG. 2A

Li-730, FIG. 2A (annotated).

Li-730's adaptive filter performs noise reduction based on the speech segment information V1.

Li-730 further discloses that its noise suppression unit suppresses the stationary and non-stationary noises “according to V1” (the speech segment

information as described in [1a]). Li-730, [0026]-[0027], FIG. 2A, 3A; Lipoff,
¶198.



Li-730, FIG. 2A (annotated).

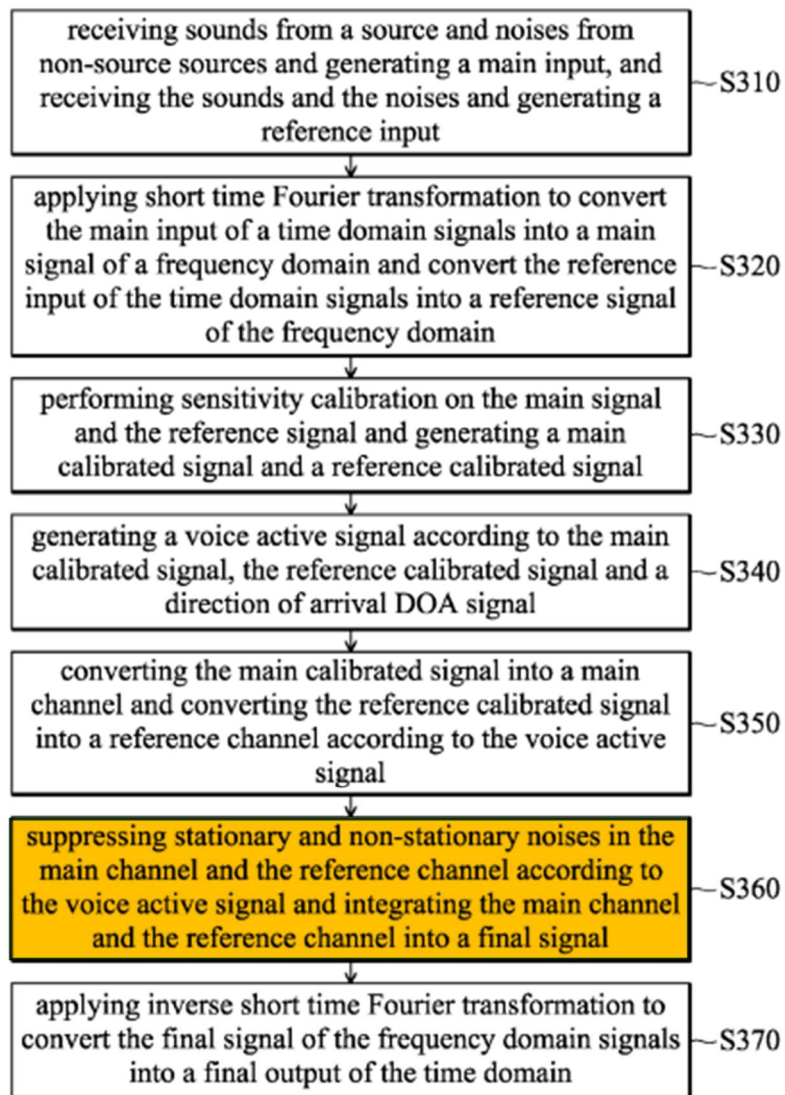


FIG. 3A

Li-730, FIG. 3A (annotated).

Li-730's adaptive filter performs noise reduction based on the voice incoming direction information N1/ N2.

Li-730's adaptive filter performs noise reduction based on the voice incoming direction information because "[t]he noise suppressing unit 250 is used

to further suppress stationary and non-stationary noises in the main channel N1 and the reference channel N2.” Li-730, [0027], FIG. 2A, 3A; Lipoff, ¶199-200. As explained in [1.c.ii] *supra*, main and reference channels (N1 and N2) include voice direction information because they are generated using “steering vector signal V2,” which “indicates [the] directional vector” of the source signal (voice signal) when the “voice active signal” is “turned on” (the power ratio of the source sound is greater than a predetermined threshold as discussed for [1a]). Li-730, [0026], [0031], claim 13, FIGS. 2A, 3C; Lipoff, ¶200.

2. Claim 12

a. [12preamble]: “An audio input apparatus comprising:”

Li-730 discloses [12preamble] because it discloses an “audio processing apparatus,” such as a cell phone, for receiving audio inputs (M1, M2) via “main” and “reference” microphones 202, 204. Li-730, [0019], [0020]; Lipoff, ¶218.

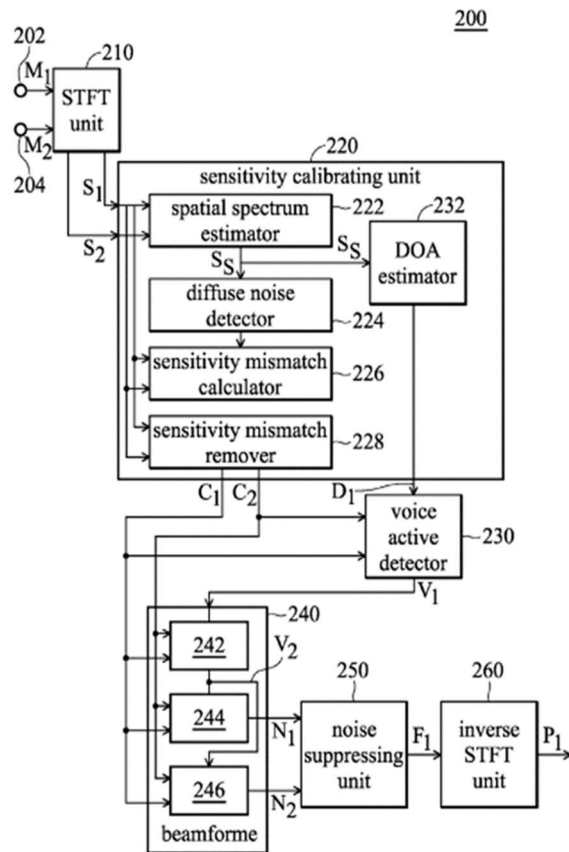


FIG. 2A

Li-730, FIG. 2B.

- b. [12a]: “a first face and an opposite second face that is apart from the first face with a specific distance;”

Li-730 discloses [12a] because the audio processing apparatus includes a “front panel 310” (also called “front side”) and an opposite “back panel 320” (also called “back side”) separated by a “physical barrier” (a specific distance). Li-730, [0020], FIG. 2B; Lipoff, ¶219.

- c. **[12b]: “a first microphone and a second microphone provided on the first face and the second face, respectively;”**

Li-730 discloses [12b] because the “main microphone” (first microphone) 202 is on the front panel and the “reference microphone” (second microphone) 204 is on the rear panel. Li-730, [0020], FIG. 2B; Lipoff, ¶220.

- d. **[12c]: “a speech segment determiner configured to determine whether or not a sound picked up by at least either the first microphone or the second microphone is a speech segment and”**

[12c] is substantively similar to [1a]. Li-730 discloses [12c] for reasons discussed for [1a], *supra*. Lipoff, ¶221.

- e. **[12d]: “to output speech segment information when it is determined that the sound picked up by the first or the second microphone is the speech segment;”**

[12d] is substantively similar to [1b]. Li-730 discloses [12d] for reasons discussed for [1b], *supra*. Lipoff, ¶222.

- f. **[12e]: “a voice direction detector configured, when receiving the speech segment information, to detect a voice incoming direction indicating from which direction a voice sound travels”**

[12e] is substantively similar to [1c]. Li-730 discloses [12e] for reasons discussed for [1c], *supra*. Lipoff, ¶223.

- g. [12e.i]: “based on a first sound pick-up signal obtained based on a sound picked up by the first microphone and a second sound pick-up signal obtained based on a sound picked up by the second microphone and”**

[12e.i] is substantively similar to [1c.i]. Li-730 discloses [12e.i] for reasons discussed for [1c.i], *supra*. Lipoff, ¶224.

- h. [12e.ii]: “to output voice incoming-direction information when the voice incoming direction is detected; and”**

[12e.ii] is substantively similar to [1c.ii]. Li-730 discloses [12e.ii] for reasons discussed for [1c.ii], *supra*. Lipoff, ¶225.

- i. [12f]: “an adaptive filter configured to perform a noise reduction process using the first and second sound pick-up signals based on the speech segment information and the voice incoming-direction information.”**

[12f] is substantively similar to [1d]. Li-730 discloses [12f] for reasons discussed for [1d], *supra*. Lipoff, ¶226.

3. Claim 13

- a. [13preamble]: “A noise reduction method comprising the steps of:”**

Li-730 discloses [13preamble] because it discloses a noise reduction method. Li-730, [0028], FIG. 3A; Lipoff, ¶227.

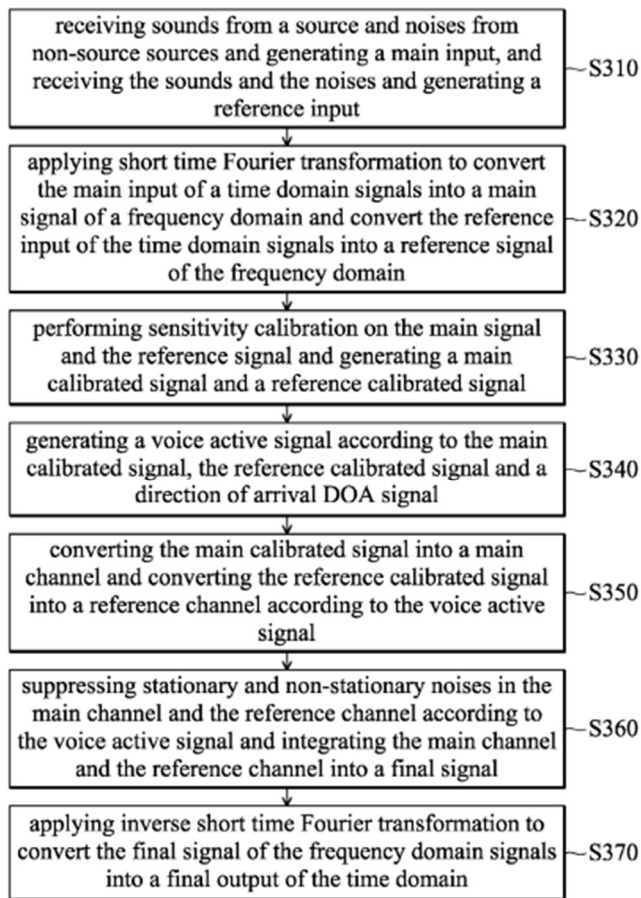


FIG. 3A

Li-730, FIG. 3A.

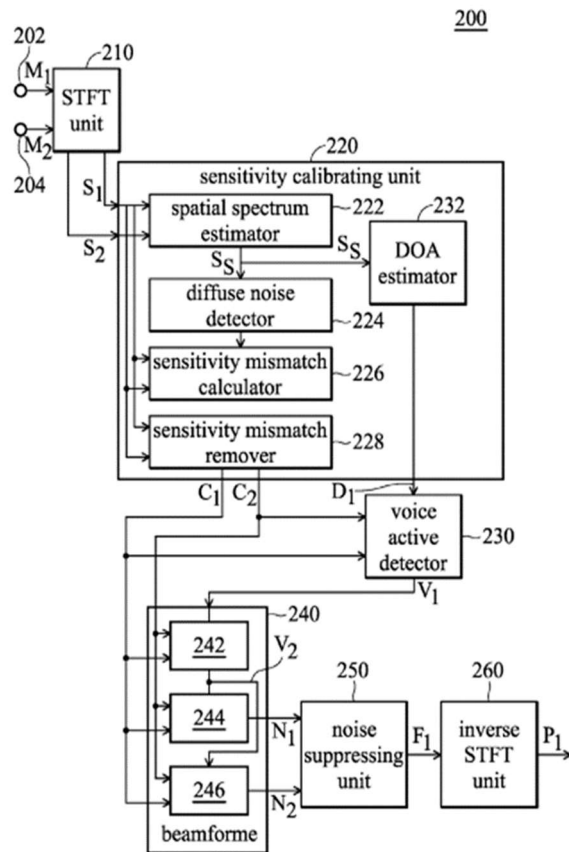


FIG. 2A

Li-730, FIG. 2B.

- b. [13a]: “determining whether or not a sound picked up by at least either a first microphone or a second microphone is a speech segment;”

[13a] is substantively similar to [1a]. Li-730 discloses [13a] for reasons discussed for [1a], *supra*. Lipoff, ¶228.

- c. **[13b]: “detecting a voice incoming direction indicating from which direction a voice sound travels, based on a first sound pick-up signal obtained based on a sound picked up by the first microphone and a second sound pick-up signal obtained based on a sound picked up by the second microphone, when it is determined that the sound picked up by the first or the second microphone is the speech segment; and”**

Li-730 discloses [13b] for reasons discussed for [1b], [1c], and [1c.i], *supra*.

Lipoff, ¶229.

- d. **[13c]: “performing a noise reduction process using the first and second sound pick-up signals based on speech segment information indicating that the sound picked up by the first or the second microphone is the speech segment and voice incoming-direction information indicating the voice incoming direction.”**

Claim [13c] is substantively similar to [1d]. Li-730 discloses [13c] for reasons discussed for [1d], *supra*. Lipoff, ¶230.

C. Dependent Claims

- 1. **Claims 2 and [14]: “the voice direction detector detects the voice incoming direction [voice incoming direction is detected] based on a phase difference between the first and second sound pick-up signals.”**

Li-730 discloses claims 2 and 14 because as explained for [1c], the beamformer (voice direction detector) uses a “steering vector” to determine the “directional vector at each frequency bin according to the voice active signal V1 which is provided by the VAD 230.” Li-730, [0026], [0031], FIGS. 2A, 3D; Lipoff, ¶¶201-202, 231.

A POSITA would have understood that the steering vector represents the set of phase-delays (amount of delay) between the calibrated sound inputs C1 and C2 (first and second sound pick-up signals) originating from the first and second microphones. Lipoff, ¶ 202. Accordingly, a POSITA would have understood that the determined directional vector for each of the first and second calibrated signals (sound pick-up signals) is based on the difference between the phases (amount of delay) of the two signals. Lipoff, ¶ 202.

2. **Claims 3 and [15]: “the adaptive filter performs the noise reduction process [the noise reduction process is performed] to reduce a noise component carried by the first sound pick-up signal using the second sound pick-up signal when the first sound pick-up signal has a more advanced phase than the second sound pick-up signal**

whereas the adaptive filter performs the noise reduction process [the noise reduction process is performed] to reduce a noise component carried by the second sound pick-up signal using the first sound pick-up signal when the second sound pick-up signal has a more advanced phase than the first sound pick-up signal.”

Li-730 discloses claims 3 and 15 because its noise suppression unit (adaptive filter) “suppress[es] stationary and non-stationary noises” (performs the noise reduction process) to reduce a noise component carried in “the main calibrated signal C1” or “reference calibrated signal C2” (the first or second sound pick-up signals) corresponding to main and reference channels N1 and N2. Li-730, [0026]-[0028]; Lipoff, ¶¶203-217, 232.

Li-730 discloses reducing a noise component carried by the first sound pick-up signal (C1) using the second sound pick-up signal (C2). Li-730, [0027]; Lipoff, ¶¶204-206.

As explained in [1d], Li-730's adaptive filter performs noise reduction using the first and second sound pick-up signals (C1, C2) because they are used to generate the main and reference channels N1 and N2 and "[t]he noise suppressing unit 250 is used to further suppress stationary and non-stationary *noises in the main channel N1* and the reference channel N2." Li-730, [0027], FIGS. 2A, 3A (emphasis added). The main channel N1 "correspond[s] to the sounds received from the source" i.e., the "speaker's voice" while reference signal N2 "correspond[s] to the noises received from non-source sources." Li-730, [0026], FIGS. 2A, 3D; *see also* [0020] (explaining that the "source" originates from "a speaker's mouth").

Thus, Li-730 discloses reducing a noise component carried by N1 corresponding to C1 (the speaker's voice) using N2 corresponding to C2 (the noises received from non-sources). Lipoff, ¶¶204-206

Li-730 discloses that C2 (second sound pick-up signal) is used to reduce the noise component in C1 (first sound pick-up signal) when C1 has an advanced phase because C1 corresponds to "source" sounds received from a front microphone that is positioned closer to the speaker's mouth. Lipoff, ¶¶207-211.

As explained in [1a], Li-730's noise reduction apparatus includes a "main microphone" disposed on a front panel of the apparatus. Li-730, [0020]. The front microphone is typically¹⁰ "closer to the source, e.g. a speaker's mouth, than the reference microphone 204" and receives sound input M1, which is calibrated into signal C1. Li-730, [0020]-[0024]; *see also* [1a]; Lipoff, ¶ 208. A POSITA would have understood that the input source sound received by the front microphone, and thus calibrated signal C1, likewise has an advanced phase relative to the non-source sounds received at the rear reference microphone, as was known in the art. Li-730, [0020]; Konchitsky, [0071] (explaining that sounds received from a "back microphone" are "delayed" relative to the "front microphone" and that the noise from the "delayed back microphone" is "subtracted" "from the front microphone signal."); Lipoff, ¶ 208.

Further, because "[t]he noise suppressing unit 250 is used to further suppress stationary and non-stationary *noises in the main channel NI*" (generated using C1

¹⁰ A POSITA would have understood that this is the case when the device is used in accordance with typical use, e.g., when the speaker is holding the apparatus and speaking into it; but, that there are alternative cases in which the rear microphone could pick up the speaker's voice first, e.g., where the speaker is not holding the device or the rear microphone is more closely aligned with the speaker's mouth.

as explained in [1d]) and to “integrate the main channel N1 and the reference channel N2 into a final signal”—that is, to suppress noises in main channel N1 using N2 when N1 contains the source sound—a POSITA would have understood that the noise suppressing unit performs the noise reduction on N1/C1 when C1 has an advanced phase, as was known in the art. Lipoff, ¶209 (citing Konchitsky, [0057] (“At block 720, the signals are delayed by one sample. At block 730, the delayed rear microphone signal is subtracted from the front microphone signal.”))).

Thus, Li-730 discloses the noise reduction apparatus according to claim 2, wherein the noise suppression unit (adaptive filter) performs the noise reduction process to reduce a noise component carried by C1 (the first sound pick-up signal) using C2 (the second sound pick-up signal) when the first sound pick-up signal is received before the second pick-up signal (has a more advanced phase than the second sound pick-up signal).

Conversely, and as explained below, a POSITA would have understood that when the rear facing microphone receives the source sound first, i.e., the second sound pick-up signal has the advanced phase, and the front microphone’s signal is delayed—the noise suppression unit reduces noise contained in C2 using the delayed non-source noise in C1.

Li-730 discloses reducing a noise component carried by second sound pick-up signal (C2) using first sound pick-up signal (C1). Li-730, [0027]; Lipoff, ¶¶ 212-217.

As discussed above, “[t]he noise suppressing unit 250 is used to further suppress stationary and non-stationary noises in the main channel N1 *and the reference channel N2.*” Li-730, [0027], FIGS. 2A, 3A (emphasis added). A POSITA would have understood that suppressing noise in the reference signal (N2 corresponding to sound pick-up signal C2) includes using the first sound pick-up signal corresponding to C1/N1. Indeed, because only two sound signals are used in Li-730’s noise reduction process (main and reference), a POSITA would have understood that the first signal is the only signal that could be used to suppress noise in N2.

Thus, Li-730 discloses reducing a noise component carried by N2 corresponding to C2 (the second sound pick-up signal) using N1 corresponding to C1 (the first sound pick-up signal).

Li-730 discloses or renders obvious that C1 (first sound pick-up signal) is used to reduce the noise component in C2 (second sound pick-up signal) when C2 has an advanced phase, because a POSITA would have understood that C2 corresponds to “source” sounds when the speaker’s mouth is closer to the second microphone. Li-730, [0027]; Lipoff, ¶¶ 215-217.

As explained above, although the front microphone is typically “closer to the source, e.g. a speaker’s mouth, than the reference microphone 204” (and receives sound input M1 which is calibrated into signal C1), a POSITA would have understood that there are alternative cases in which the rear microphone could pick-up the speaker’s voice first, e.g., where the speaker is not holding the device or the rear microphone is more closely aligned with the speaker’s mouth.

In such situations, the source (speaker) sound M2 corresponding to C2 would have an advanced phase, and noise components in the C2/N2 signal would be suppressed using C1/N1 with the objective of enhancing the sound of the voice in the C2/N2 signal. Li-730, [0027]. That is, when the rear facing microphone receives the source sound first, i.e., the second sound pick-up signal has the advanced phase, and the front microphones signal is delayed—the noise suppression unit reduces noise contained in C2 using the delayed non-source noise in C1, as was known in the art. *See, e.g.*, Konchitsky, [0057] (“The delayed front microphone signal is subtracted from the rear microphone signal.”).

X. Discretionary Denial Is Not Warranted

A. 35 U.S.C. § 314(a) Does Not Warrant Discretionary Denial

Apple Inc. v. Fintiv, Inc., IPR2020-00019, Paper 11 at 5-6 (Mar. 20, 2020) (precedential), and the Office’s interim guidelines dated June 21, 2022 (“Guidance Memo”), do not warrant discretionarily denying institution.

Petitioner is defendant in the parallel litigation captioned *SoundClear Technologies LLC v. Google LLC*, 2-24-cv-00321 (E.D. Va. May 1, 2024). The *Fintiv* factors do not favor institution because no trial date has been set in the parallel litigation and the litigation is in its infancy. Furthermore, Petitioner was diligent in filing this petition, the parties have made minimal investment in the parallel proceeding, and any overlap in the issues at this stage is speculative at best because the parties have not exchanged final contentions. Accordingly, the *Fintiv* factors favor institution and the Board should not exercise its discretion to deny institution.

B. 35 U.S.C. § 325(d) Does Not Warrant Discretionary Denial

The Board should not exercise its discretion to deny institution under § 325(d) because none of the references cited in this Petition were previously presented to or evaluated by the Examiner. *See generally* Ex. 1002. Moreover, Kale and Li-730 disclose the allegedly allowable subject matter corresponding to [1d] identified by the Examiner during prosecution. Ex. 1002, 35-36. Thus, the asserted Grounds show the Examiner erred in allowing the claims.

XI. Mandatory Notices

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

Google LLC is the real party-in-interest for this Petition.¹¹

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

To the best of Petitioner's knowledge, the '259 patent is or has been involved in the following case:

Name	Number	Court	Filed
<i>SoundClear Technologies LLC v. Google LLC</i>	2-24-cv-00321	E.D. Va.	May 1, 2024
<i>SoundClear Technologies LLC v. Amazon.com, Inc. et al.</i>	2-24-cv-00320	E.D Va.	May 1, 2024

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

Petitioner identifies the following lead and backup counsel:

Lead Counsel	Backup Counsel
Erika H. Arner, Reg. No. 57,540 erika.arners@finnegan.com Finnegan, Henderson, Farabow, Garrett & Dunner, LLP 1875 Explorer Street Reston, VA 20190-6023 Tel: 571-203-2754 Fax: 202-408-4400	Cory C. Bell (Reg. No. 75,096) cory.bell@finnegan.com Finnegan, Henderson, Farabow, Garrett, & Dunner LLP 2 Seaport Lane, 6th Floor Boston, MA 02210-2001 Tel.: 617-646-1641 Fax: 202-408-4400

¹¹ Google LLC is a subsidiary of XXVI Holdings Inc., a subsidiary of Alphabet Inc. XXVI Holdings Inc. and Alphabet Inc. are not real parties-in-interest here.

Lead Counsel	Backup Counsel
	Michael V. Young (Reg. No. 61,180) michael.young@finnegan.com Finnegan, Henderson, Farabow, Garrett & Dunner, LLP 1875 Explorer Street Suite 800 Reston, VA 20190-6023 Tel: 571-203-2788 Fax: 202-408-4400 Safiya Aguilar (Reg. No. 77,212) safiya.aguilar@finnegan.com Finnegan, Henderson, Farabow, Garrett & Dunner, LLP 901 New York Avenue, NW Washington, DC 20001-4413 Tel: 202-408-4160 Fax: 202-408-4400

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

Please address correspondence to lead and backup counsel at the addresses above and Google-Soundclear-IPRs@finnegan.com. Petitioner consents to electronic service by e-mail.

XII. Standing

Petitioner certifies that the '259 patent is available for *inter partes* review and Petitioner is not barred or estopped from requesting *inter partes* review on the grounds herein.

XIII. Conclusion

Petitioner has established a reasonable likelihood of prevailing and requests the Board institute *inter partes* review and cancel each challenged claim.

Date: February 10, 2025

Respectfully submitted,

/Erika H. Arner/

Erika H. Arner

Lead Counsel

Reg. No. 57,540

CERTIFICATE OF COMPLIANCE

The undersigned hereby certifies that the foregoing **Petition for *Inter Partes* Review** contains 13,861 words, excluding those portions identified in 37 C.F.R. § 42.24(a), as measured by the word-processing system used to prepare this paper.

/Erika H. Arner/
Erika H. Arner
Lead Counsel
Reg. No. 57,540

CERTIFICATE OF SERVICE

The undersigned certifies that the foregoing **Petition for *Inter Partes* Review of U.S. Patent No. 9,031,259, the associated Power of Attorney, and Exhibits 1001 through 1014** were served on February 10, 2025, by FedEx Priority Overnight® on the correspondence address of record indicated in the Patent Office's Patent Center system for U.S. Patent No. 9,031,259.

Kevin Sprenger
DAIGNAULT IYER LLP
8229 Boone Boulevard
Suite 450
Vienna, VA 22182

Date: February 10, 2025

/Daniel E. Doku/
Daniel E. Doku
Senior Litigation Legal Assistant
FINNEGAN, HENDERSON, FARABOW,
GARRETT & DUNNER, LLP