

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

SAMSUNG ELECTRONICS CO., LTD.,
SAMSUNG ELECTRONICS AMERICA, INC.,

Petitioners

v.

STATON TECHIYA, LLC,

Patent Owner

Case IPR2024- 01033

U.S. Patent No. 11,750,965

Petition for *Inter Partes* Review of

U.S. Patent No. 11,750,965

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PETITIONERS' EXHIBIT LIST

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1001	U.S. Patent 11750,965 (“’965pat”)
1002	Declaration of Dr. Kyriakakis (“Kyriakakis”)
1003	Curriculum Vitae of Dr. Kyriakakis
1004	File History of U.S. Patent 11,750,965
1005	US Patent Pub. 20140270200A1 (“Usher-1”)
1006	US Patent Pub. 20200380945A1 (“Woodruff-1”)
1007	US Patent 11,393,486 (“Woodruff-2”)
1008	US Patent Pub. 20140126748A1 (“Usher-2”)
1009	US Patent Pub. 20110299695 (“Nicholson”)
1010	Redline comparing the ’965Pat to its parent (Patent No. 11,277,682 (“’682Pat”))
1011	U.S. Patent No. 11,277,682
1012	U.S. Provisional Application 60/893,617
1013	Notice of Allowance in U.S. Application 14/109,987
1014	Redline comparing the Patent No. 11,277,682 (’682Pat).to Patent No. 8,625,812 (“’812Pat”)
1015	Letter from Nikhil Krishnan to Thomas J Friel, Jr.

LISTING OF CHALLENGED CLAIMS

Claim 1	
1.pre	An earbud comprising:
1.1	a first Ambient Sound Microphone (ASM), wherein the first ASM generates a first ASM signal;
1.2	a second ASM, wherein the second ASM generates a second ASM signal;
1.3	an ear canal microphone (ECM), wherein the ECM generates an ECM signal;
1.4	a speaker;
1.5	a memory that is configured to store instructions; and
1.6	a processor that executes the instructions to perform operations, wherein the processor is coupled to the first ASM, wherein the processor is coupled to the second ASM, wherein the processor is coupled to the ECM, wherein the processor is coupled to the speaker, and the operations comprising:
1.7	receiving the first ASM signal;
1.8	receiving the second ASM signal;
1.9	receiving the ECM signal;
1.10	generating a first signal from at least one of the first ASM signal, or the second ASM signal, or both;

1.11	generating an ambient sound signal from the first signal or a modified first signal;
1.12	detecting if the user is speaking by analyzing at least one of the first signal, or the first ASM signal, or the second ASM signal, or the ECM signal, or by analyzing the first signal and the ECM signal; and
1.13	adjusting at least one of the ambient sound signal, an audio content signal, a mixed ambient sound signal and audio content signal, a noise signal or a combination thereof, when it is detected that the user is speaking.
Claim 2	
2	The earbud according to claim 1 wherein the operation of adjusting, adjusts the audio content signal by reducing its volume.
Claim 3	
3	The earbud according to claim 2 wherein the operation of adjusting also amplifies the ambient sound signal.
Claim 4	
4	The earbud according to claim 1 wherein the operation of adjusting, adjusts the ambient sound signal by increasing its volume.
Claim 5	
5	The earbud according to claim 1 wherein the operations further comprise: If the user speaking is detected, analyzing at least one of the first ASM signal, the second ASM signal, the ECM signal, or a combination thereof to detect a voice command.
Claim 6	
6	The earbud according to claim 3 wherein the operations further comprise: resetting the volume of the ambient sound signal to a preset value when a preset time increment has expired.

Claim 7	
7	The earbud according to claim 4 wherein the operations further comprise: resetting the volume of the ambient sound signal to a preset value when a preset time increment has expired.
Claim 8	
8	The earbud according to claim 3 wherein the operations further comprise: resetting the volume of the audio content signal to a preset value when a preset time increment has expired.
Claim 9	
9	The earbud according to claim 1 wherein the operations further comprise: determining the background noise level by analyzing at least one of the first ASM signal, the second ASM signal, the ECM signal, or a combination thereof.
Claim 10	
10	The earbud according to claim 1 wherein the operations further comprise: determining the background noise level by analyzing at least one of a modified first ASM signal, a modified second ASM signal, modified ECM signal, or a combination thereof.
Claim 11	
11	The earbud according to claim 9, wherein the operations further comprise: generating an anti-noise signal using at least one of the first ASM signal, the second ASM signal, the ECM signal or a combination thereof.
Claim 12	

12	The earbud according to claim 10, wherein the operations further comprise: generating an anti-noise signal using at least one of the modified first ASM signal, the modified second ASM signal, the modified ECM signal or a combination thereof.
Claim 13	
13	The earbud according to claim 11, wherein the operations further comprise: mixing the ambient sound signal and audio content signal to generate a mixed signal.
Claim 14	
14	The earbud according to claim 13, wherein the operations further comprise: generating a new mixed signal by combining the mixed signal with the anti-noise signal.
Claim 15	
15	The earbud according to claim 14, wherein the operations further comprise: reducing the anti-noise signal and increasing the ambient sound signal prior to mixing if the user's voice is detected.
Claim 16	
16	The earbud according to claim 12, wherein the operations further comprise: mixing the ambient sound signal and audio content signal to generate a mixed signal.
Claim 17	
17	The earbud according to claim 16, wherein the operations further comprise:

	generating a new mixed signal by combining the mixed signal with the anti-noise signal.
Claim 18	
18	The earbud according to claim 17, wherein the operations further comprise: reducing the anti-noise signal and increasing the ambient sound signal prior to mixing if the user's voice is detected.
Claim 19	
19[pre]	An earbud comprising:
19.1	a first Ambient Sound Microphone (ASM), wherein the first ASM generates a first ASM signal;
19.2	a second ASM, wherein the second ASM generates a second ASM signal;
19.3	an ear canal microphone (ECM), wherein the ECM generates an ECM signal;
19.4	a speaker;
19.5	a memory that is configured to store instructions; and
19.6	a processor that is coupled to the first ASM, wherein the processor is coupled to the second ASM, wherein the processor is coupled to the ECM, wherein the processor is coupled to the speaker,
19.7	wherein the processor receives the first ASM signal, wherein the processor receives the second ASM signal, wherein the processor receives the ECM signal,
19.8	wherein the processor generates a first signal from at least one of the first ASM signal, or the second ASM signal, or both,

19.9	wherein the processor generates an ambient sound signal from the first signal or a modified first signal,
19.10	wherein the processor detects if a user is speaking by analyzing at least one of the first signal, or the first ASM signal, or the second ASM signal, or the ECM signal, or by analyzing the first signal and the ECM signal, and
19.11	adjusts a level of the ambient sound signal sent to the speaker if the user's voice is detected.
Claim 20	
20	The earbud according to claim 19, wherein an audio content sent to the speaker is adjusted if the user's voice is detected.
Claim 21	
21[pre]	A method comprising:
21.1	receiving the first ASM signal from a first ambient sound microphone;
21.2	receiving the second ASM signal from a second ambient sound microphone;
21.3	receiving the ECM signal from an ear canal microphone;
21.4	generating a first signal from at least one of the first ASM signal, or the second ASM signal, or both;
21.5	generating an ambient sound signal from the first signal or a modified first signal;
21.6	detecting if the user is speaking by analyzing at least one of the first signal, or the first ASM signal and the second ASM signal, or the first ASM signal and the ECM signal, or the second ASM signal and the ECM signal; and

21.7	adjusting a mixed signal sent to the speaker, wherein the mixed signal is a combination of the ambient sound signal, the audio content signal, and an anti-noise signal, when it is detected that the user is speaking.
Claim 22	
22	The method according to claim 21 further comprising: adjusting the mixed signal by reducing a volume of the audio content signal.
Claim 23	
23	The method according to claim 21 further comprising: adjusting the mixed signal by increasing a volume of the ambient sound signal.
Claim 24	
24	The method according to 31 further comprising: analyzing at least one of the first ASM signal, the second ASM signal, the ECM signal, or a combination thereof to detect a voice command.
Claim 25	
25	The method according to claim 23 further comprising: resetting the volume of the ambient sound signal to a preset value when a preset time increment has expired.
Claim 26	
26	The method according to claim 22 further comprising: resetting the volume of the audio content signal to a preset value when a preset time increment has expired.
Claim 27	
27	The method according to claim 21 further comprising:

	generating a background noise level by analyzing at least one of the first ASM signal, the second ASM signal, the ECM signal, or a combination thereof.
Claim 28	
28	The method according to claim 21 further comprising: generating a background noise level by analyzing at least one of a modified first ASM signal, a modified second ASM signal, a modified ECM signal, or a combination thereof.
Claim 29	
29	The method according to claim 21 further comprising: generating the anti-noise signal using at least one of the first ASM signal, the second ASM signal, the ECM signal or a combination thereof.
Claim 30	
30	The method according to claim 21 further comprising: generating the anti-noise signal using at least one of the modified first ASM signal, the modified second ASM signal, the modified ECM signal or a combination thereof

I. INTRODUCTION

Samsung Electronics Co., Ltd., Samsung Electronics America, Inc. (together “Samsung”) (collectively, “Petitioners”) request *inter partes* review (“IPR”) of claims 1-30 (“Challenged Claims”) of Patent No. 11,750,965 (“the ’965pat”).

II. GROUNDS FOR STANDING

Petitioners certify that the ’965pat is available for IPR and that Petitioners are not barred or estopped from requesting IPR.

III. STATEMENT OF PRECISE RELIEF REQUESTED FOR EACH CLAIM CHALLENGED

Petitioners respectfully request review and cancellation under 35 U.S.C. §311 as follows:¹

Ground	Claims	Basis ²
1	1-4, 6-9, 11, 13-15, 19-23, 25-27, 29	§103: <i>Usher-1-Woodruff-1-Woodruff-2</i>
2	5, 24	§103: <i>Usher-1-Woodruff-1-Woodruff-2-Usher-2</i>

¹ Petitioners do not concede that any Challenged Claims satisfy other requirements for patentability that cannot be raised in IPR.

² As discussed below, Petitioners’ obviousness grounds do not require the bodily incorporation of components from one reference into another, but instead are directed to the application of the teachings of one reference to another.

Ground	Claims	Basis ²
3	10, 12, 16-18, 28, 30	§103: <i>Usher-1-Woodruff-1-Woodruff-2-Nicholson</i>

As shown below, each reference pre-dates the earliest priority date to which the Challenged Claims may be entitled, which is no earlier than June 22, 2021.

Reference	Date	AIA Art ³ At Least Under
US20140270200A1 (“ <i>Usher-1</i> ”)	09/18/2014 (published)	§102(a)(1)
US20200380945A1 (“ <i>Woodruff-1</i> ”)	10/13/2016 (published)	§102(a)(1)
US11,393,486B1 (“ <i>Woodruff-2</i> ”)	05/13/2020 (filed)	§102(a)(2)
US20140126748A1 (“ <i>Usher-2</i> ”)	05/08/2014 (published)	§102(a)(1)
US20110299695A1 (“ <i>Nicholson</i> ”)	12/08/2011 (Published)	§102(a)(1)

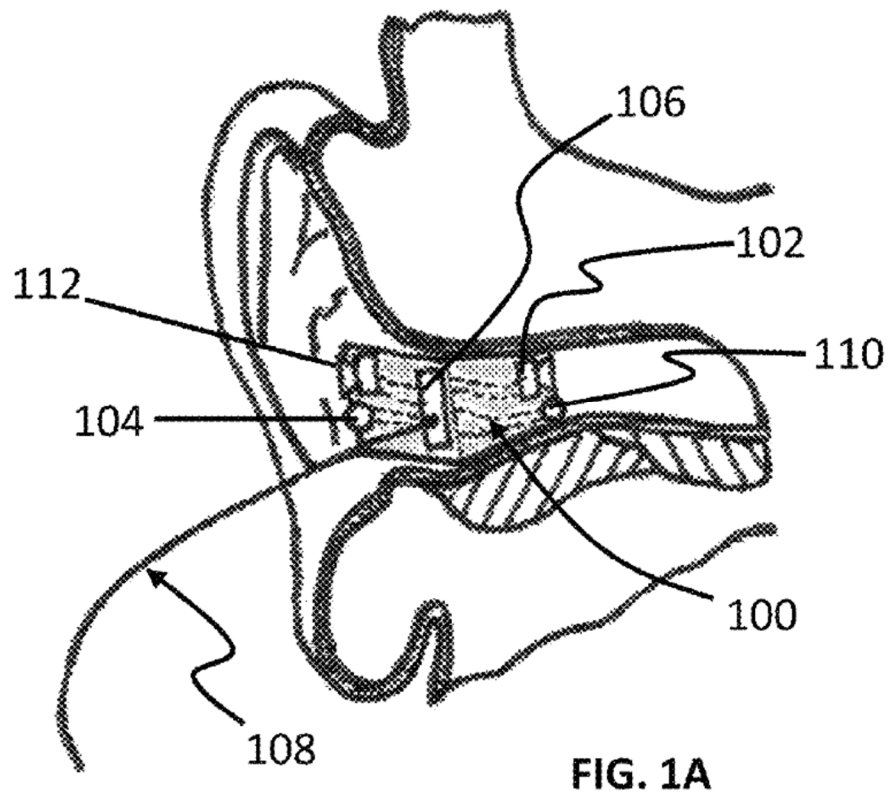
IV. OVERVIEW OF THE ‘965pat

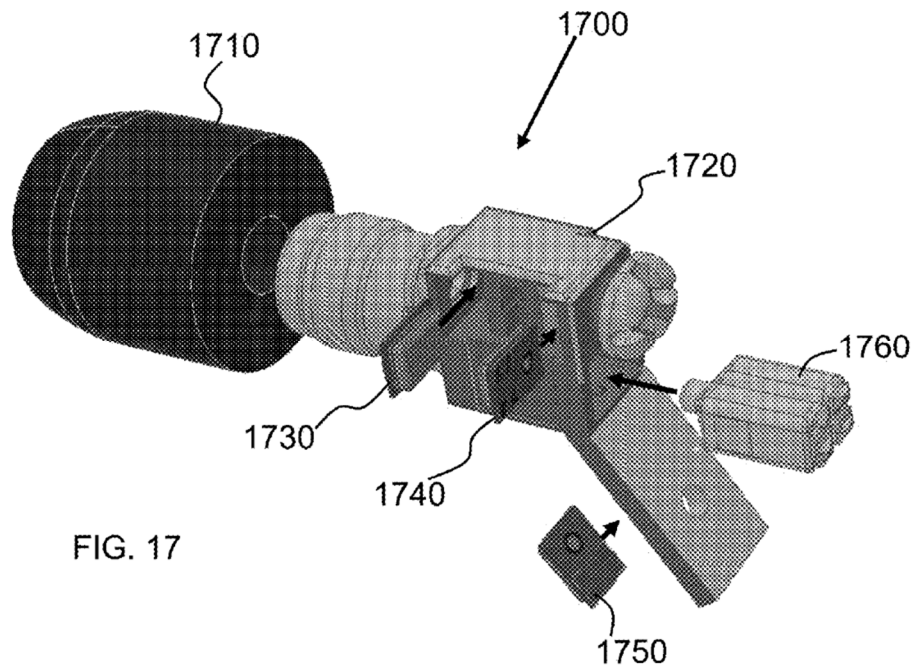
A. The Specification

The ‘965pat discloses a system that allows acoustic detecting and/or measuring devices to compensate for the adverse effects caused by “acoustic dampening.” ‘965pat, 1:23-37. While the ‘965pat states that the apparatus/method described therein can be used for various applications, the specification focuses on

³ The ‘965pat patent is subject to the AIA because its claims are not entitled to the benefit of any of the applications from which it claims priority. See AIA §3(n)1(A).

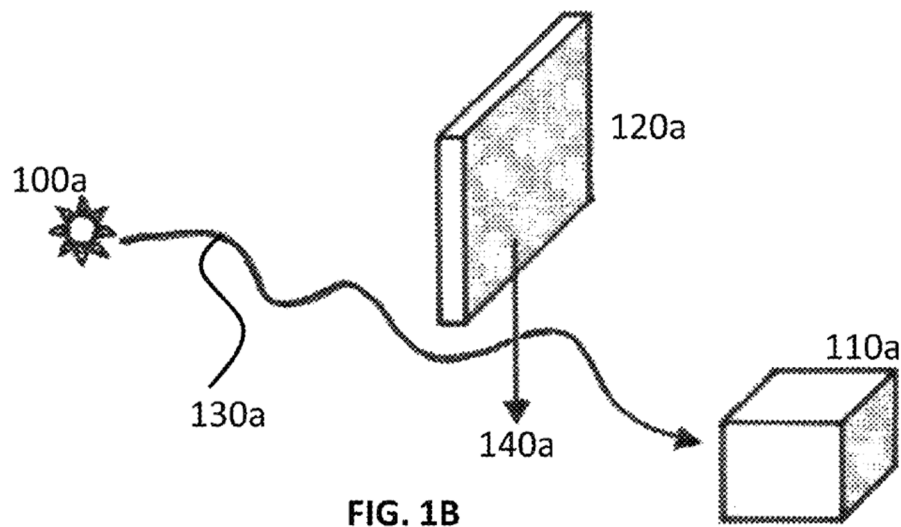
earphone applications. *See e.g., id.* 3:39-43; Figs. 1, 7, 9-14 and 17-19. The components of exemplary earphone embodiments described in the '965*pat* are seen in Figs. 1A and 17:





The '965pat's earphone has ambient sound microphones ("ASMs") (e.g., Fig. 1A's ASMs 104 and Fig. 17's ASMs 1740). As shown, the ASMs are located outside of the wearer's ear canal and positioned to detect ambient sound. '965pat, 4:7-9; 28:24-30. The '965pat's earphone has an ear canal microphone ("ECM") (e.g., Fig. 1A's ECM 110 and Fig. 17's ECM 1730). The ECM is located at the user's ear canal. *Id.* 4:13-20; 28:24-30. The '965pat's earphone has loudspeaker (e.g., Fig. 1A's loudspeaker 102 and Fig. 17's speaker 1760). The loudspeaker is positioned to deliver audio to the wearer. *Id.* 4:5-7; 28:24-30.

The '965pat discloses that acoustic dampening events occur when an acoustic dampening element moves into an acoustic signal/wave's path. *Id.* 4:21-27. Fig. 1B shows this:

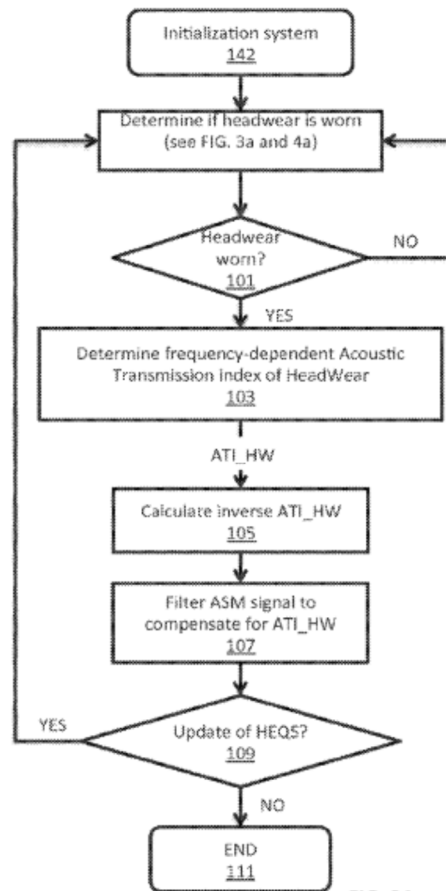


Acoustic dampening element 120a moves into the path of an acoustic signal/wave 130a generated by an acoustic source 100a. The '965pat states that acoustic dampening element 120a can dampen acoustic signal/wave 130a, affecting the acoustic signal measured by microphone 110a. *Id.* 4:21-27.

Despite boilerplate language asserting that the patent has many applications, all embodiments relate to methods allowing earphones to respond to acoustic dampening events, such as a wearer putting a hat on their head that occludes the ASM(s) detecting ambient sound. Indeed, according to the '965pat's Field Of The Invention, the subject of the patent is "acoustic compensation of acoustic dampening by headwear on detected acoustic signals." *Id.* 1:23-26. The '965pat's Summary Of The Invention is similar. *See id.* 1:51-66; 2:16-27. The specification recites the word "headwear" 107 times.

In fact, the specification refers to the technology described therein as a “headwear acoustic equalization system,” or “HEQS.” Essentially every embodiment is described as an HEQS system. *See e.g., Id.* 4:28-38; 4:56-62; 5:6-11; 5:47-51; 6:8-14; 6:42-43; 6:59-67; 7:24-28; 8:26-9:29; 9:30-10:29; 10:30-11:48; 11:49-12:47; 12:48-13:63.

Fig. 2A showing an HEQS method for determining whether headwear is worn, and if so, how the HEQS responds:



When HEQS determines that headwear is worn (step 101), it determines what the '965Pat refers to as the “frequency dependent acoustic transmission index of the headwear,” or “ATI_HW” (step 103). The HEQS then calculates the inverse of ATI_HW (“inverse ATI_HW”) (step 105). The signal generated by the ASM then filters with the inverse ATI_HW signal (step 107). *Id.* 4:28-46. According to the '965Pat, this filtering “gives a modified ASM signal which approximates that the ASM signal with the headwear removed.” *Id.* 4:46-48. *See Kyriakakis*, ¶¶38-51.

B. Prosecution History

The '965Pat was filed on June 22, 2021 as a continuation-in-part of Application 16/662,505 (“’505App”), which issued as Patent No. 11,277,682. Substantial new matter was added to this CIP, including Figs. 7-20 and all the subject matter found at 13:64-29:32.⁴ The additions to the specification can be seen in the comparison between the '965Pat and its parent, Patent No. 11,277,682 (“’682Pat”). *See EX1010*. Despite filing the application as a continuation-in-part in 2021, which added substantial new matter, Patent Owner (“PO”) did not check the box on the Application Data Sheet indicating that the application should be examined as an AIA application. *EX1004* at 1005. The Examiner never raised this issue, and the application was examined as a pre-AIA application.

⁴ In this IPR, this new matter is called “CIP New Matter.”

The '682*Pat*, and thus the '965*Pat*, claims priority to the application, filed on March 7, 2008, that issued as Patent No. 8,625,812 ("'812*Pat*"), through a series of continuation applications. The '965*Pat* also claims the benefit of Provisional Application 60/893,617 ("'617*Prov*").⁵

The '965*Pat* was filed with 12 claims, one of which was independent. None of the originally-filed claims were directed to an earphone that detects acoustic damping events. *EX1004* at 1038-1040. Instead, as-filed claim 1 claimed a "voice pickup system" having several elements, and recited the functionality of "detect[ing] if the user is speaking." *Id.*

On 02/10/2022, PO filed preliminary amendment that amended claim 1, cancelled claims 2-11 and added claims 12-28. PO amended claim 1 to recite "adjusting at least one of the ambient sound signal, an audio content signal, a noise signal, or a combination thereof, when it is detected that the user is speaking." *Id.* at 898-901. The following new claims were added:

- Claims 16-18 (issued claims 6-8) recited "resetting the volume" of various signals "when a preset time increment has expired."

⁵ Information found in both the original specification and the '617*Priv*, i.e., '965*Pat*, Figs. 1-6 and 1:1-13:63 is referred to as "Original Disclosure."

- Claims 19-20 (issued claims 9-10) recited “determining the background noise level” by analyzing particular signals.
- Claims 21-22 (issued claims 11-12) recited “generating an anti-noise signal” by using particular signals.
- Claims 23 and 26 (issued claims 13 and 26) recited “combining the ambient sound signal and audio content signal to generate a mixed signal.”
- Claims 24 and 27 (issued claim 14 and 17) recited “generating a new mixed signal by combining the mixed signal with the anti-noise signal.”
- Claims 25 and 28 (issued as claim 15 and 18) recited “reducing the anti-noise signal and increasing the ambient sound signal prior to mixing if the user’s voice is detected.”

In the 02/10/2022 preliminary amendment, PO asserted that the amendments were supported by Figs. 2B and 19, as well as paragraphs [0042], [0045], [0048], [0049], [0055], [0058], [0072], [0077], [0084], [0089], and [0112] *Id.* at 897. Fig. 19 and paragraphs [0072], [0077], [0084], [0089], and [0112] are CIP New Matter. *Kyriakakis*, ¶57.

On 09/29/2022, an office action issued rejecting all claims for non-statutory double patenting. *Id.* at 333-339.⁶ On 12/12/2022, PO responded by filing a Terminal Disclaimer. *Id.* at 323. In the listing of claims, two minor amendments are highlighted using standard PTO amendment notations. *Id.* at 319. In its Remarks, PO stated that “Claim 1 has been amended for grammatical issues.” *Id.* at 323. This statement, however, was inaccurate, as the “adjusting...” element contained an additional limitation that had not previously been considered by the Examiner. Had the proper amendment notations been made, the Examiner would have seen the following:

adjusting at least one of the ambient sound signal, an audio content signal, a mixed ambient and audio signal, a noise signal, or a combination thereof, when it is detected that the user is speaking.⁷

The Examiner issued a Notice Of Allowance on 02/10/2023. *Id.* at 282-288. In the Notice of Allowance, the Examiner referred to comments 1-3 in the Notice of Allowance issued in Application 14/109,987 (*See Id.* at 287).⁸ Comment 1

⁶ Application 10/123,110 is one of the parent applications in the '965*pat.*'s chain of priority.

⁷ All **bold**/*italics*/underline added unless noted otherwise.

⁸ The 14/109,987 application is in the priority chain of the '965*Pat.*

discusses a distinction made in during prosecution of that application regarding claim language directed to acoustic damping, which is not recited in any of the claims of the '965Pat. See EX1013 at 6. Comment 2 noted that the Examiner was construing the processor element to be limited to the methods shown in Figs. 2a, 3A-3B, 4A-4B and 6 (which are also in the '965Pat). *Id.* Comment 3 referred to examiner amendments made in the 14/109,987 application, not the '965Pat. *Id.*

On 02/20/2023, PO filed an amendment requesting claim amendments be entered “to correct grammatical and antecedent issues.” EX1004 at 269-273. On 03/06/2023, the Examiner entered the amendments. EX1004 at 264-265.

On 05/10/2023, PO filed an RCE to add method claims “that are similar to the allowed device claims. *Id.* at 257. Although PO stated that the purpose was to add method claims, PO also added additional apparatus claims (application claims 29-30). On 06/05/2023, the Examiner issued an office action rejecting all claims for being indefinite because “it is not clear how to read ‘combination thereof’ as recited, noting that the first ASM signal, the second ASM signal, the ECM signal are the signals received from respective microphones. There are no means contemplated to combine the analog signals from each microphone.” EX1004 at 207. The Examiner again referred to comments 1-3 made in the 14/109,987 Notice Of Allowance. *Id.* at 206-297.

On 06/07/2023, PO responded to the 06/05/2023 Office Action by making additional amendments to claims 1, 29 (issued claim 19), and 31 (issued claim 21). Claim 1 was amended to add “generating a first signal from at least one of the first ASM signal, or the second ASM signal.” *Id.* at 193. Claim 1 was further amended to recite that the “ambient sound signal” is generated from the “first signal or a modified first signal.” *Id.* Finally, claim 1 was amended to recite that the process of detecting if the user speaking is performed “by analyzing at least one of the first signal, or the first ASM signal, or the second ASM signal, or the ECM signal, or by analyzing the first signal and the ECM signal.” *Id.* (emphasis in original). In Remarks, PO stated that “the present application claims priority to Provisional Application No. 60/893,617 filed on 7 Mar. 2007. The invention claimed herein predates 16 March 2013 and is entitled and requested to be examined under Pre-AIA provisions.” *Id.* at 199. PO also indicated that Fig. 20 and Paragraphs [0106] and [0112] described combining multiple ASM signals. *Id.* at 200. Fig. 20 and Paragraphs [0106] and [0112] were CIP New matter.

On 07/17/2023, the Examiner issued a Notice Of Allowance that once again once again referred to comments 1-3 made in the 14/109,987 Notice Of Allowance. *Id.* at 32. The Examiner also indicated that “[r]eceiving the first ASM, second ASM and ECM signals are drawn to the functions defined in fig. 2b.” *Id.* The Examiner stated this even though Fig. 2b shows neither an ECM nor any ECM

signals. Neither does the portion of the specification discussing Fig. 2B. *'965Pat*, 4:56-5:5. The Examiner indicated that the claims were deemed allowable because “[i]n the examiner’s opinion, it would not have been obvious to a person of ordinary skill in the art to implement the claimed signal processing in the prior art communication devices using a filter to compensate for an acoustic environment around a microphone.” *Id.* at 33. Respectfully, the Examiner stated this even though no claims recite any filtering, and no functionality is recited regarding compensation for the acoustic environment around a microphone.

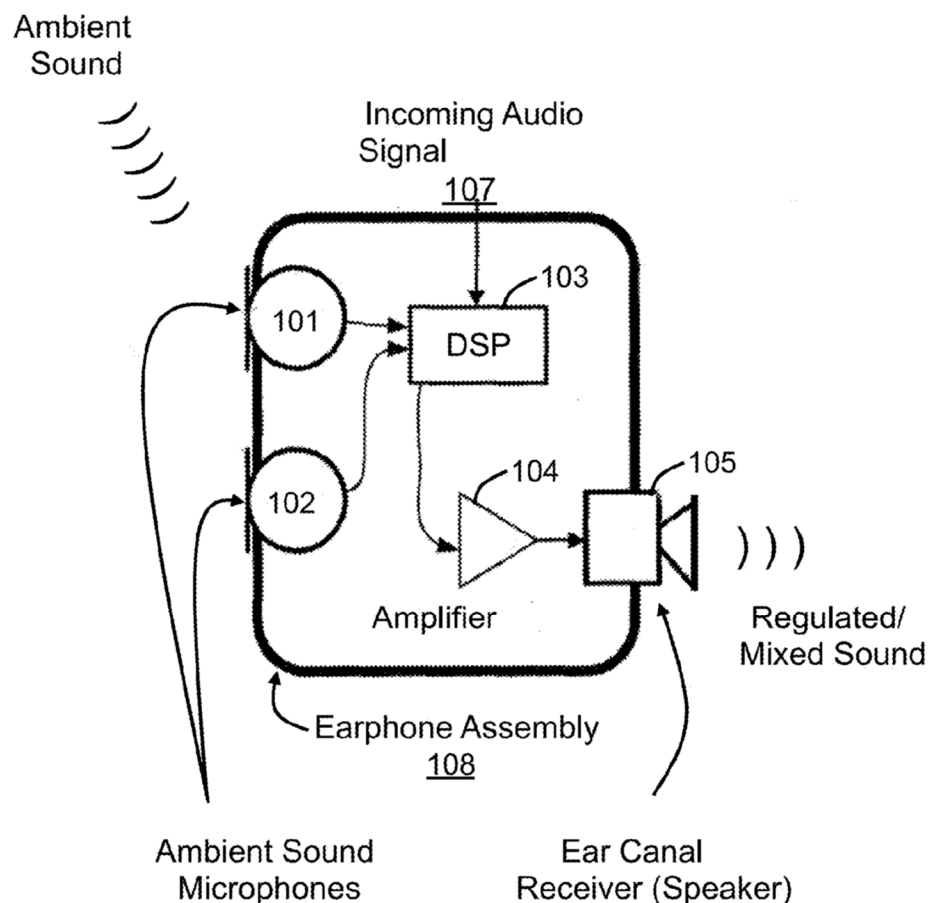
V. OVERVIEW OF THE PRIOR ART

The prior art references analyzed in the grounds below are analogous to the *'965pat* because they are within the same field of endeavor and/or reasonably pertinent to one or more problems addressed by the *'965pat*. Like the *'965pat*, *Usher-1*, *Woodruff-1/Woodruff-2*, *Usher-2*, and *Nicholson* disclose earphone embodiments for analyzing the acoustic signals and responding to those detected signals with various functionality. *See, e.g.,* Abstracts in *Usher-1*, *Woodruff*, *Usher-2*, *Nicholson*,. *Kyriakakis*, ¶68.

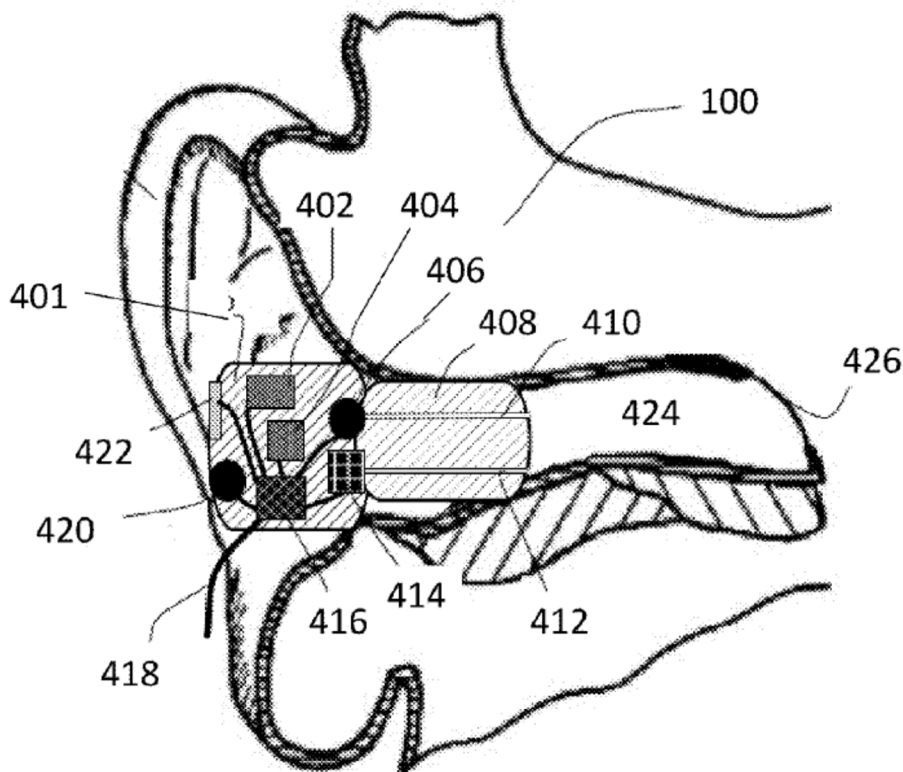
A. *Usher-1*

Usher-1 is titled “System And Method To Detect Close Voice Sources And Automatically Enhance Situation Awareness.” *Usher-1* discloses “a method and system for automatically detecting when an earphone user and a second nearby user

are engaged in a voice conversation, and activating a sound pass-through system such that the nearby voice source is automatically heard by the user of the sound isolating earphone.” *Usher-I*, ¶[0019]. *Usher-I* discloses that “Fig. 1A illustrates an earphone 100 that can provide sound isolation and is configured by way of electronic components to detect close voice sources and enhance situation awareness...” *Id.* ¶[0020]. Fig. 1A shows earphone 100 having ambient sound microphones 101/102, loudspeaker 105, and processor 107.



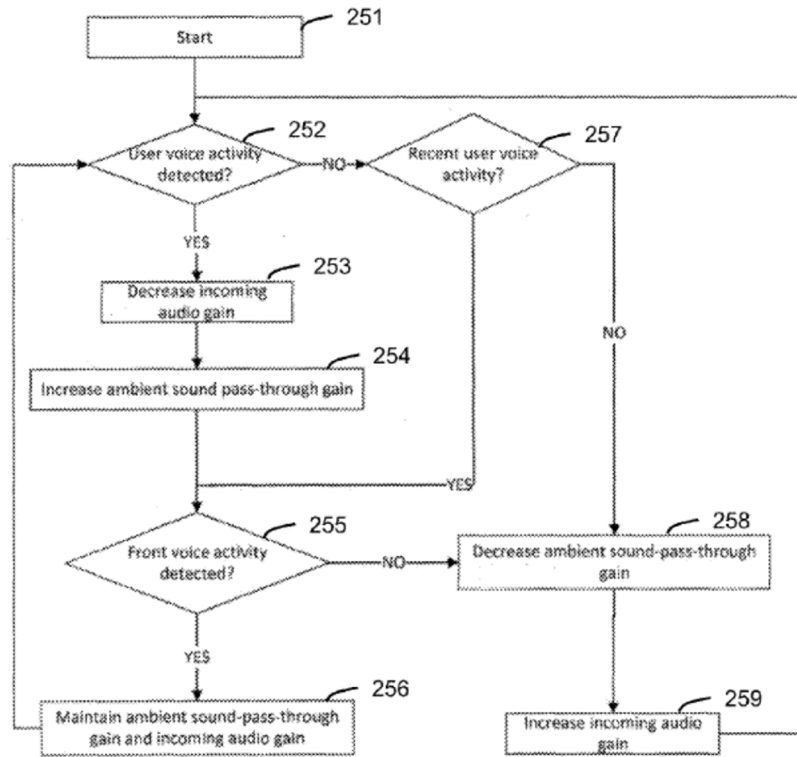
Usher-1, Fig. 1A; ¶[0052]. *Usher-1*'s Fig. 4 shows its earphone inserted into a user's ear:



400
FIG. 4

Usher-1's ¶[[[0055] states that the earphone shown in Fig. 4 has “microprocessor 416, memory 404, battery 402, ECM 406, ASM 420, ECR, 414, and user interface 422.” *Id.* *Usher-1* further discloses that microprocessor 416 “is operatively coupled to memory 404, ECM 406, ASM 420, ECR 414, and user interface 420.” *Id.*

Usher-1's Fig. 2B is a flowchart showing *Usher-1*'s ambient sound pass-through method:



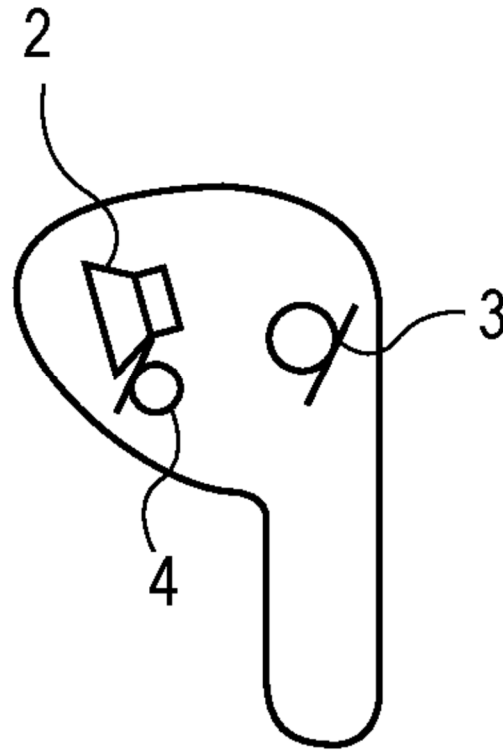
250
FIG. 2B

See also *Usher-1*, ¶[0029]. *Usher-1*'s processor monitors sound from its ASMs and detects whether voice is detected. *Id.* at ¶[0030]. At 252, if voice is detected, the gain of the audio signal being played back (e.g., music) is reduced (step 253) while the gain of the signals from *Usher-1*'s ASMs is increased (step 254). *Id.* See *Kyriakakis*, ¶¶69-73.

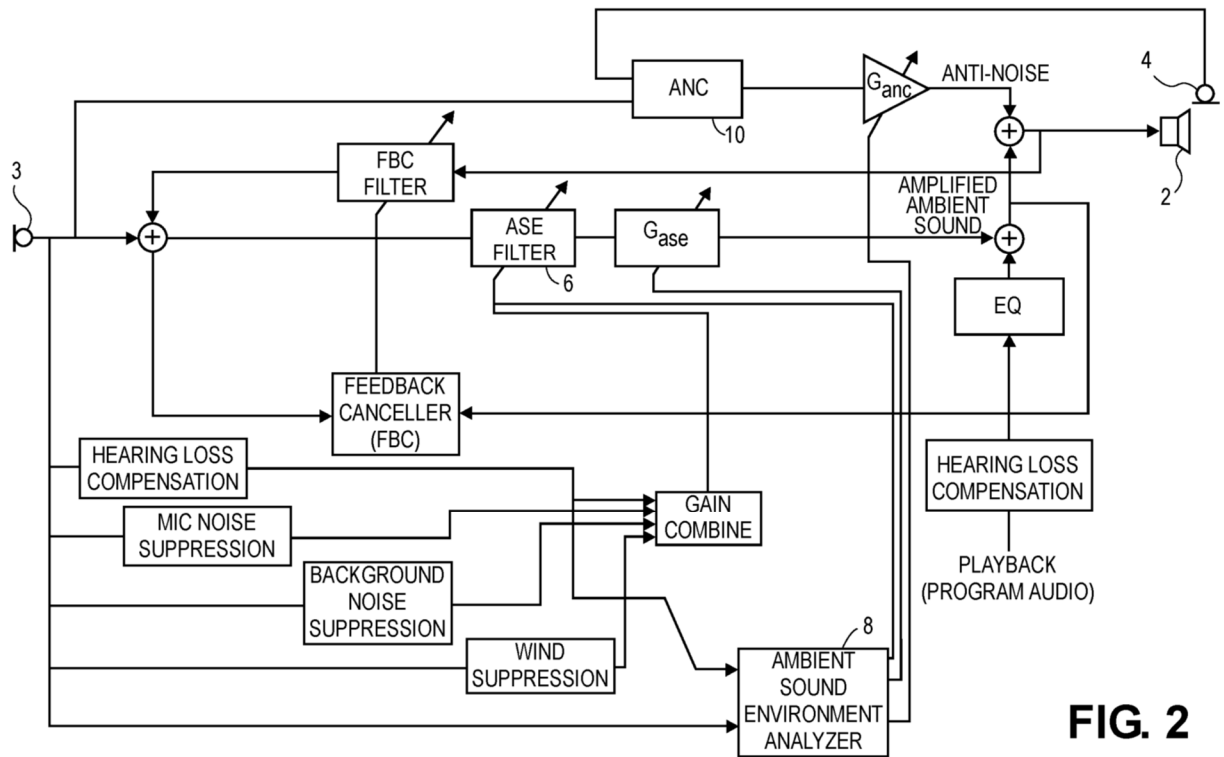
B. *Woodruff-1*

Woodruff-1 is titled “Ambient Sound Enhancement And Acoustic Noise Cancellation Based On Context.” *Woodruff-1* discloses “an audio system that has an ambient sound enhancement function, also referred to here as a transparency mode of operation, in which an against-the-ear audio device having one or more speakers converts a digitally processed version of an input audio signal into sound.” *Woodruff-1*, ¶[0004]. POSITAs would have understood that transparency mode for earphones entails reproducing both audio content (e.g., music) while also reproducing sound from the ambient environment. *Id.* See also *Kyriakakis*, ¶75. According to *Woodruff-1*, “[r]eproduced ambient sound may be louder as compared to the user hearing the ambient sound without the device against their ear...” *Woodruff-1*, ¶[0004]. See also *id.*, ¶[0025].

Like *Usher-1*, *Woodruff-1* discloses an earphone (i.e., an against-the-ear audio device) having an ASM 3, and ECM 4, and a loudspeaker 2. See *Woodruff-1*, ¶[0020]. These components are shown in the portion of Fig. 1 reproduced below:



Woodruff-1's Fig. 2 details the manner in which signals generated by ASM 3 and ECM 4 are modified and processed by its signal processing circuitry to implement the disclosed transparency mode:

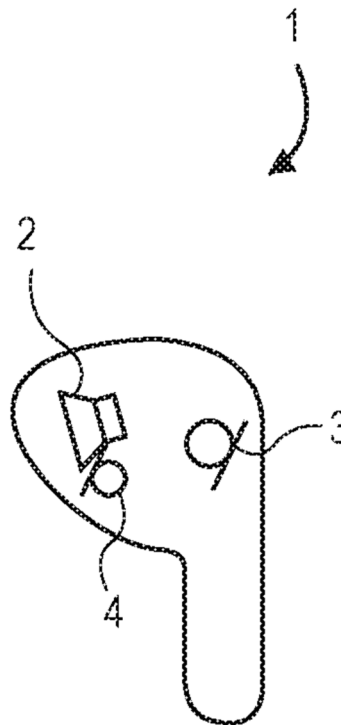


As can be seen, ASM 3 detects ambient sound, converts the ambient sound to electrical signals, which are filtered by ASE filter 6. The signal output by ASE filter 6 are input to gain block G_{ase} , which outputs an amplified ambient sound signal. The amplified ambient sound signal is then summed with program audio, and this summed signal is then combined with an anti-noise signal. Then, the combined audio (i.e., program audio and ambient sound) are played back via speaker 2. See Kyriakakis, ¶¶74-78.

C. *Woodruff-2*

Woodruff-2 is titled “Ambient Noise Aware Dynamic Range Control And Variable Latency For Hearing Personalization.” *Woodruff-2* discloses noise-aware

dynamic range controller for an earphone device. *Woodruff-2*, 1:28-34. Like *Woodruff-1*, *Woodruff-2* has an ASM 3, an ECM 4 and an in-ear speaker 2, as shown in Fig. 1:



Woodruff-2's Fig. 3 shows a “block diagram of an audio system and related method for personalized ASE...” *Id.*, 3:7-9. The device shown *Woodruff-2*'s Fig. 3 includes what POSITAs would have understood is a feedback cancellation circuit, including a “Feedback Canceller” and “FBC filter,” just as shown in *Woodruff-1*'s Fig. 2:

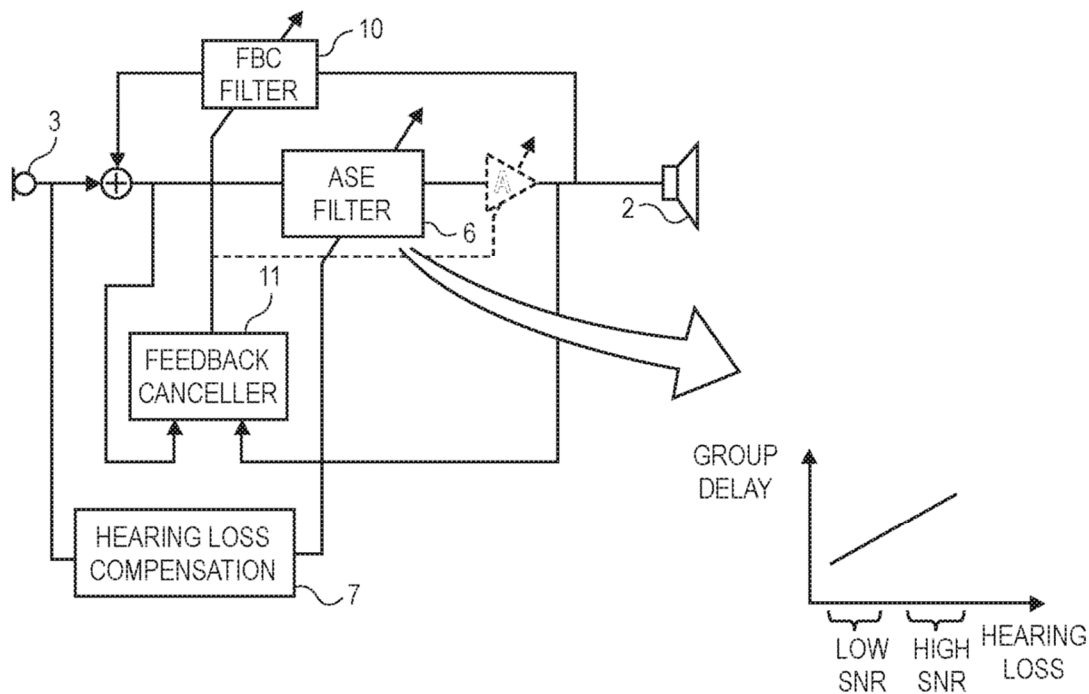
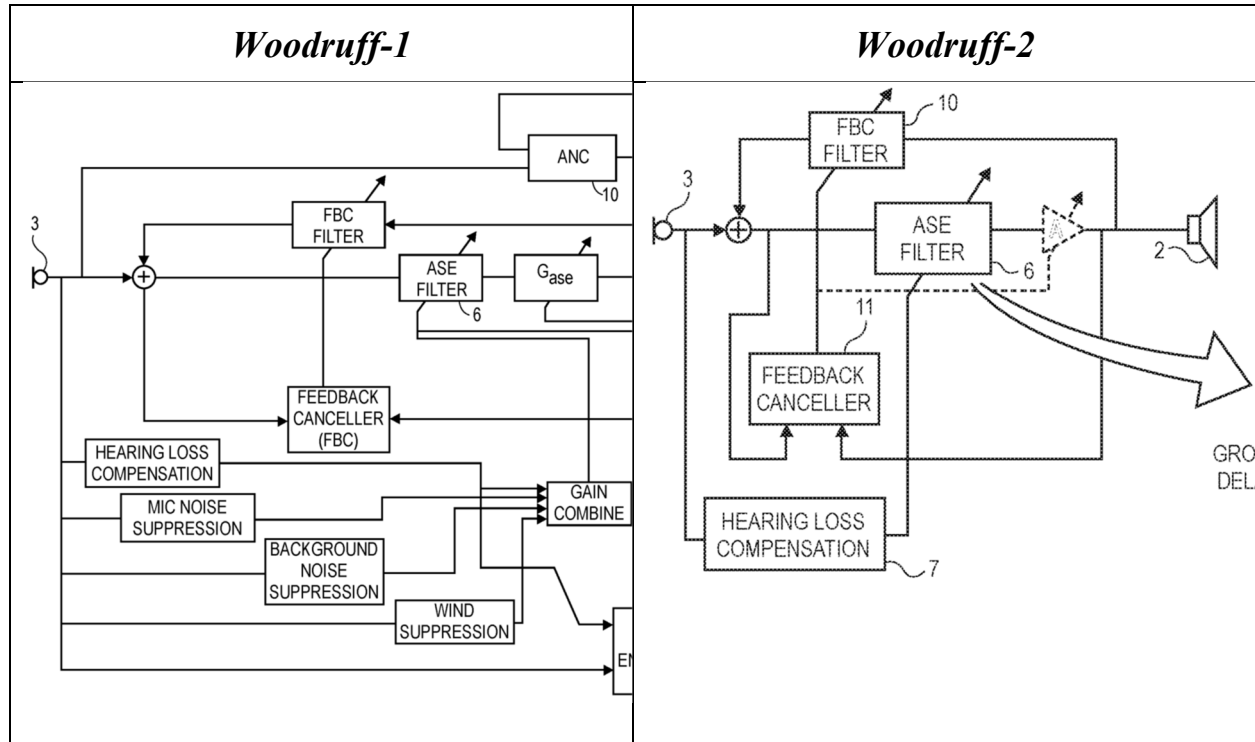


FIG. 3

As *Woodruff-2* states, “ASE filter 6 filters the audio signal coming from the external microphone 3.” *Woodruff-2*, 6:61-62. *Woodruff-2* further discloses that its ASE filter 6 “appl[ies] a feedforward gain” to the audio signal. *Id.*, 6:64-66. *Woodruff-2* also discloses that feedback cancellation can also be applied to the audio signal from microphone 3. *Woodruff-2* does this by filtering the output of ASE filter 6 via FBC filter 10. *Id.*, 7:21-28. Then, “[t]he output of the FBC filter 10 may be added to input audio signal coming from the external microphone 3 to result in a combined signal at the input of the ASE filter 6.” *Id.*, 7:33-36.

Note that POSITAs would have recognized that *Woodruff-2*’s feedback cancellation circuit shown in Fig. 3 is used in *Woodruff-1*’s Fig. 2, which can be seen

by comparing the feedback cancellation circuits of each side-by-side:



Woodruff-1, Fig. 2 (excerpt); *Woodruff-2*, Fig. 3 (excerpt). See also See Kyriakakis, ¶¶79-83.

D. *Usher-2*

Usher-2 is titled “Method And Device For Voice Operated Control.” Two of the three listed inventors are the same as listed on *Usher-1*, and *Usher-1* incorporates *Usher-2* by reference. *Usher-1*, ¶[0050]. *Usher-2* discloses “[m]ethods and devices for wearable sound processing and voice operated control...” *Usher-2*, Abstract. *Usher-2* discloses an earphone having a processor that analyze signals for information “such as voice commands.” *Usher-1*, ¶[0050]. Kyriakakis, ¶84.

E. *Nicholson*

Nicholson is titled “Active Noise Cancellation Decision In A Portable Audio Device.” *Nicholson* discloses “Active noise cancellation (ANC) circuitry is coupled to the input of an earpiece speaker in a portable audio device, to control the ambient acoustic noise outside of the device and that may be heard by a user of the device.”

Nicholson, Abstract. Fig. 2 shows *Nicholson*’s components:

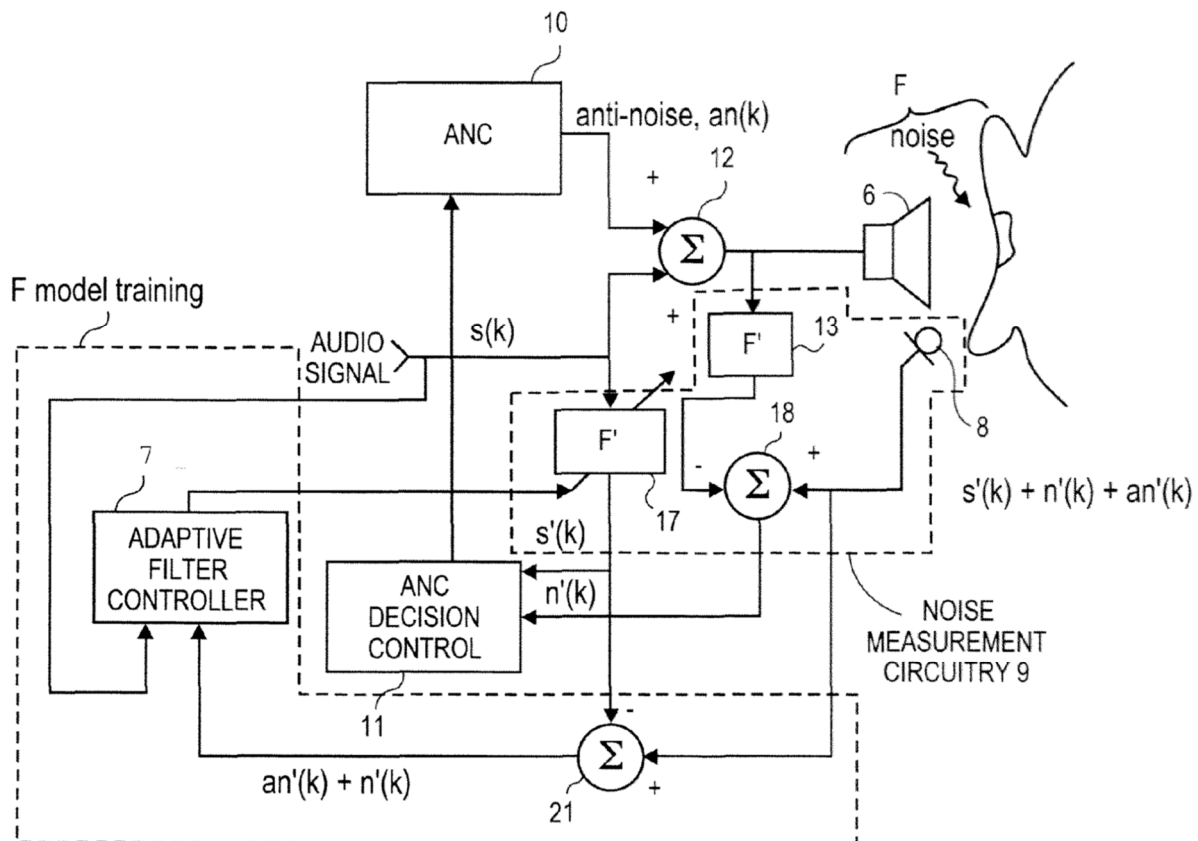


FIG. 2

As is seen, *Nicholson* “includes an error microphone 8 that is located and oriented in such a manner as to pickup both (a) sound emitted from the earpiece speaker 6

and (b) the ambient acoustic noise that has leaked into the cavity or region between the handset housing or shell (not shown) that is in front of the earpiece speaker 6 and the user's ear.” *Nicholson*, ¶[0027]. *Nicholson* uses various signal processing techniques, discussed in more detail below, to attenuate ambient background noise. *Kyriakakis*, ¶85.

VI. LEVEL OF ORDINARY SKILL IN THE ART

Persons of ordinary skill in the art at the time of the alleged invention of the '965pat (“POSITA”) would have had a bachelor’s degree in electrical engineering, computer science, audio engineering, or a similar field and at least two years of experience with hearing instruments such as earphones, hearing aids and/or hearing protectors. POSITAs could have also obtained similar knowledge and experience through other means. *Kyriakakis*, ¶22.

VII. CLAIM CONSTRUCTION

Petitioners are unaware of any “prior claim construction determination” by any tribunal. *See* 37 C.F.R. §42.100(b).

Here, given the close correlation and substantial identity between the prior art references and the Challenged Claims, Petitioners submit that no express constructions of the claims are necessary, and the claims should be given their

ordinary meaning.⁹ *Phillips v. AWH Corp.*, 415 F.3d 1303, 1312-13 (Fed. Cir. 2005) (en banc).

**VIII. THE CHALLENGED CLAIMS ARE NOT ENTITLED TO THE
BENEFIT OF EITHER APPLICATION 12/044,727 OR '617Prov**

Claims 1, 19 and 21 are each independent. Challenged claims 2-18 depend on claim 1, challenged claim 20 depends on claim 19, and challenged claims 22-30 depend on claim 21. Each independent claim recites limitations that are not supported by either the ultimate parent application (application 12/044,727, (“’727App”)) or the ’617Prov, and in some cases, even the specification of the ’965Pat. Thus, no claims have a priority date any earlier than 06/22/2021.

As discussed, the ’965Pat was filed as a CIP of the ’505App”. The ’505App claimed priority to the ’727App through a chain of continuations. All applications in the priority chain prior to the ’965Pat contain essentially the same specification, with minor changes made to the Summary Of Invention. *See EX1014*. (comparing the ’682Pat to the ’812Pat). The following summarizes the limitations that are

⁹ Petitioners reserve all rights to raise claim construction arguments and other arguments in any parallel or future litigation concerning the ’965Pat. For example, comparing the claims to the accused products in the litigation may raise controversies that require construction of certain claim terms.

unsupported by the Original Disclosure, or the '965*Pat* itself:

Claims	Claim Element
1, 2, 8, 20, 22, and 26	“Adjusting” or “resetting” an “audio content signal.”
1, 13-18, and 21-23	“Mixing” an ambient sound signal with an audio content signal and/or an anti-noise signal.
6-8 and 25-26	Resetting volume of a signal after a “preset time increment.”
9-12, 14-15, 17-18, 21, 27- 30	“Anti-noise signal” and “Background noise level.”
1, 19, and 21	“Detecting if the user is speaking by analyzing . . . the ECM signal.”

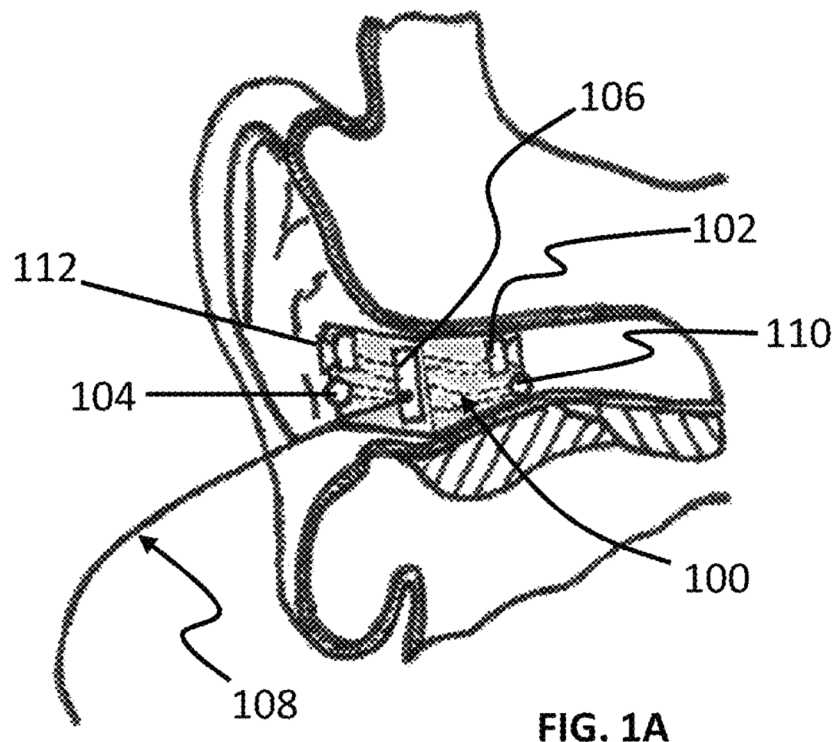
As discussed in Section IV.A, the applications in the '965*Pat*'s priority chain are generally directed to a “Headwear Acoustic Equalization System (HEQS)” that can be used to compensate an ASM signal during an “acoustic dampening event,” such as headwear placed over an earpiece microphone.

The '965*Pat*'s claim language is divorced from the priority applications. The claims do not include limitations related to “headwear” or detecting a “dampening event,” —i.e., the core concepts disclosed in the priority applications. Instead, the claims are fundamentally directed to an earbud that “adjusts” one or more signals when it is detected that the user is speaking.

A. “Adjusting” or “resetting” an “audio content signal.”

There is no support in the Original Disclosure for the element “adjusting ... an audio content signal,” as required by claim 1. Instead, the only support is found in CIP New Matter added when the ’965*Pat* was filed, meaning that claims reciting this limitation are entitled to a priority date no earlier than 06/22/2021, the ’965*Pat*’s filing date. *Parus Holdings, Inc. v. Google LLC*, 70 F.4th 1365, 1374 (Fed. Cir. 2023). *See also Kyriakakis*, ¶¶95-104.

Neither the Original Disclosure nor the CIP New Matter ever use the term “audio content signal.” The Original Disclosure, however, discloses that its earphone receives “an audio signal” being input at connector 108, which is delivered to the user via loudspeaker 102. ’965*Pat*, 4:4-13. *See* Fig. 1A:



POSITAs would understand that the terms “audio signal” and “audio content signal” may be synonyms. *Kyriakakis*, ¶96. Regardless, there is no disclosure of adjusting such an audio signal in the Original Disclosure, let alone adjusting it in response to detecting that the user is speaking. All references in the Original Disclosure to an audio signal are as follows: 4:7, 4:12-13, 8:62-63, 9:62-64, 10:12, 11:1-3, 11:17, 12:7-9, and 13:13-15.¹⁰ There is no mention of adjusting anywhere.

¹⁰ Note that the Original Disclosure also refers to an “audio signal filter.” *See* 7:14-18 and 8:32:38. However, the Original Disclosure only discloses applying an audio

In contrast, the CIP New Matter discloses that its device “adjust[s] audio content.” *'965Pat*, 17:64-18:4. *See also Id.*, 22:48-54. Thus, any written description support for this claim element is found only in the CIP New Matter, meaning claim 1 (and claims dependent therefrom) is not entitled to a priority date earlier than the *'965Pat*'s filing date.

Claim 2 recites adjusting audio content signal “by reducing its volume.” This is not supported by any disclosure and thus not entitled to rely on its claimed priority date.

Claim 8, which depends (through claims 2 and 3) on claim 1 recites, in part, “resetting the volume of the audio content signal to a preset value.” Neither the term “resetting” nor any root thereof appears in the *'965Pat*. There are no synonyms either, meaning that this claim is unsupported by the *'965Pat*. Thus, claim 8 is not entitled to rely on the claimed priority date.

Claim 20 recites that “an audio content sent to the speaker is adjusted if the user's voice is detected.” Thus, claim 20 only finds written description support, if

signal filter to ambient sound signals, which are not audio content signals. *Id.* The Original Disclosure also refers to a SONAR test audio signal, but there is no disclosure of adjusting such a signal, and regardless, SONAR is not “audio content.” *Id.*, 10:20-25 and 11:26-31.

any, in the CIP New Matter, and is therefore not entitled to a priority date any earlier than the CIP filing date.

Dependent claim 22 recites “adjusting the mixed signal by reducing a volume of the audio content signal.” As discussed, only the CIP New matter discloses adjusting the audio content. However, no disclosure in *'965Pat* disclosing “reducing the volume” of the audio content signal when it is detected that the user is speaking, meaning that claim 22 is not entitled to its claimed priority date. .

Finally, claim 26 recites “resetting the volume of the audio content signal to a preset value...” As discussed, “resetting” the audio content signal is not disclosed at all. Thus, claim 26 is not entitled to its claimed priority date.

B. “Mixing” an ambient sound signal with an audio content signal and/or an anti-noise signal

Claims 1, 13-18, and 21-23 recite limitations related to mixing an ambient sound signal with an audio content signal and/or an anti-noise signal. The following is the specific claim language which is unsupported by the Original Disclosure:

Claim	Claim Language
1	“mixed ambient sound signal and audio content signal”
13, 16	“mixing the ambient sound signal and audio content signal to generate a mixed signal”
14, 17	“generating a new mixed signal by combining the mixed signal with the anti-noise signal”
15, 18	“prior to mixing”
21	“adjusting a mixed signal sent to the speaker, wherein the mixed signal is a combination of the ambient sound signal, the audio content signal, and an anti-noise signal”
22-23	“adjusting the mixed signal”

Neither the term “mix,” “mixed” nor “mixing appear anywhere in the specification of the ’965*Pat.* The only description that might be considered mixing ambient sound and audio content signals is found in CIP New Matter, for example at 16:29-37. *See also id.*, 21:23-30. Note that the CIP New Matter states that the ambient sound microphones can capture “audio content occurring in an environment that the earphone device 2415 is present in and/or is proximate to.” *Id.*, 18:8-17. POSITAs would consider signals created from the captured “audio content occurring in an environment” to correspond to the claimed “ambient sound signal.” *Kyriakakis*, ¶107. POSITAs would further consider signals carrying music or telephone calls to correspond to the claimed “audio content signal.” *Id.* Thus, when the CIP New Matter refers to “a combination thereof,” this is the only disclosure that

a POSITA would have considered as supporting claims reciting “mixing” an audio content signal and an ambient sound signal. Since this disclosure only appears in the CIP New Matter, claims 1, 13, 16 are not entitled to a priority date any earlier than the CIP filing date of 06/22/21.

Claims 14 and 17 each require “mixing” the previously mixed ambient sound and audio content signal with an anti-noise signal. Claims 15 and 18 depend, from claims 14 and 17, respectively, and recite “reducing the anti-noise signal ... prior to mixing...” As noted in Section IV.B, these claims were added to the application via the 02/10/2022 preliminary amendment. As discussed in Section VIII.D, there is no disclosure in either the Original Disclosure or the CIP New Matter of generating an anti-noise signal, meaning there can be disclosure of mixing an anti-noise signal with any other signals. Thus, these claims are not entitled to rely on the filing dates of either the *'617Prov* or any of the utility applications in the *'965Pat*'s priority chain. *Parus*, 70 F.4th at 1374.

Independent claim 21 recites that the “mixed signal” is a combination of various signals and an “anti-noise signal.” Claims 22-23 recite “adjusting” components of this mixed signal. As noted in Section IV.B, these claims were added to the application in the 05/10/2023 RCE. Because there is no disclosure in either the *'617Prov*, the Original Disclosure or the CIP New Matter of an anti-noise signal (*see* Section VIII.D), there can be disclosure of a mixed signal that includes an anti-

noise signal or making any adjustments to components of such a mixed signal. Thus, claims 21-23 are not entitled to rely on the filing dates of either the '617*Prov* or any of the utility applications in the '965*Pat*'s priority chain. *Parus*, 70 F.4th at 1374. *See also Kyriakakis*, ¶¶105-110.

C. Resetting volume of a signal after a “preset time increment.”

Claims 6-8 and 25-26 recite resetting the volume of either the ambient sound or the audio content signal “after a preset time increment.” As noted in Section IV.B, claims 6-8 were added to the application via the 02/10/2022 preliminary amendment, while claims 25-26 were added via the 05/10/2023 RCE. Neither the Original Disclosure nor the CIP New Matter disclose resetting either the ambient sound or audio content signals “after a present time increment.” As discussed in Section VIII.A, there is no disclosure in any of the priority applications of “resetting” audio content signals. However, while the CIP New Matter discloses adjustment of audio content signals, it provides no disclosure of exactly what it means to perform such an adjustment. This is a problem in and of itself, but even if POSITAs could extrapolate some suggestion somewhere in the '965*Pat* that such “adjustment” teachings includes resetting the volume, there is no disclosure at all of doing so “after a preset time increment.” Thus, Claims 6-8 and 25-26 are not entitled to rely on either the filing dates of the '617*Prov* or any of the utility applications in the '965*Pat*'s priority chain. *Parus*, 70 F.4th at 1374. *See also Kyriakakis*, ¶¶111-112.

D. “Anti-noise signal” and “Background noise level.”

Claims 9-10 and 27-28 recite “generating a background noise signal.” Claims 11-12, 14-15, 17-18, 21 and 29-30 recite an “anti-noise signal.” Neither the Original Disclosure nor the CIP New Matter disclose either a background noise level or an anti-noise signal, let alone use of particular inputs used to generate such signals. While the Original Disclosure contains the word “noise,” there is no mention at all of either a “background noise level” or an “anti-noise signal.” *See* '965Pat, 4:61-62 (“local noise sources, such as machine noise”), 5:65 (“nose bursts”), and 10:13/11:19 (“Noise signal (white noise or pink noise).”). Indeed, the patent states that “noise reduction” is “not described here.” '965Pat, 4:13-20. *See also Kyriakakis*, ¶113.

Thus, Claims 9-12, 14-15, 17-18, 21, 27- 30 are not entitled to rely on the filing dates of either the '617Prov or any of the utility applications in the '965Pat's priority chain. *Parus*, 70 F.4th at 1374.

E. “Detecting if the user is speaking by analyzing . . . the ECM signal.”

Claims 1, 19, and 21 each require detecting if a user is speaking by analyzing at least one of a variety of different signals, including an ECM Signal. Neither the Original Disclosure nor the CIP new matter disclose detecting if the user is speaking by analyzing an “ECM signal,” i.e., a signal generated by an ear canal microphone. As discussed in Section IV.A, the embodiments of the '965Pat describe detecting when headwear is placed on a user's head, and enabling a compensation filter that

processes the audio signals played back to the user via the earphone's loudspeaker, which is located in the user's ear canal. The '965Pat further discloses that this compensation can be voice activated. '965Pat, 2:28-29 and 1:64-66.

The only signals the '965Pat discloses are used to detect whether a user is speaking are ASM signals, i.e., signals generated by ambient sound microphones. One of these embodiments is described at 11:48-12:47, and it states that a "Voice test sound is generated by the HEQS user, and is simultaneously detected using one or more ambient sound microphones mounted on the same headset." *Id.*, 11:50-53. Another embodiment, described at 12:48-13:63, also uses only signals generated by ambient sound microphones to detect whether a user is speaking. *Id.*, 12:52-58 ("Once obtained, the ASM signal is filtered with a filter corresponding to the inverse of ATI_HW. This self-contained Voice-based headwear determination platform uses a Voice or non-voice (e.g. hand-clap) test sound created by the HEQS user, and is simultaneously recorded using one or more ambient sound microphones mounted on a headset near to or in the user's ear canal."). *See also id.*, 28:31-36.

The only disclosure regarding ear canal microphones is that they are used to detect sound in the ear canal. *Id.*, 28:66-67 ("Microphone 2020 can be used to detect audible sound in the ear canal (ear canal microphone).). There is no suggestion that signals generated by the ECM can be used to detect whether the user is speaking. The '965Pat provides no details regarding the purpose of the ear canal microphone,

other than to state that it (and an external loudspeaker) exists “for performing other functions of the headphone system not described here (such as monitoring of sound exposure and ear health conditions, headphone equalization, headphone fit testing, noise reduction, and customization).” *Id.*, 4:13-30. *See also Kyriakakis*, ¶¶115-118.

Thus, Claims 1, 19, and 21 are not entitled to rely on the filing dates any applications in the '965 *Pat*'s priority chain. *Parus*, 70 F.4th at 1374.

IX. DETAILED EXPLANATION OF GROUNDS

A. Ground 1: *Usher-1*, *Woodruff-1*, and *Woodruff-2* Render Obvious Claims 1-4, 6-9, 11, 13-15, 19-23, 25-27, 29

1. Claim 1

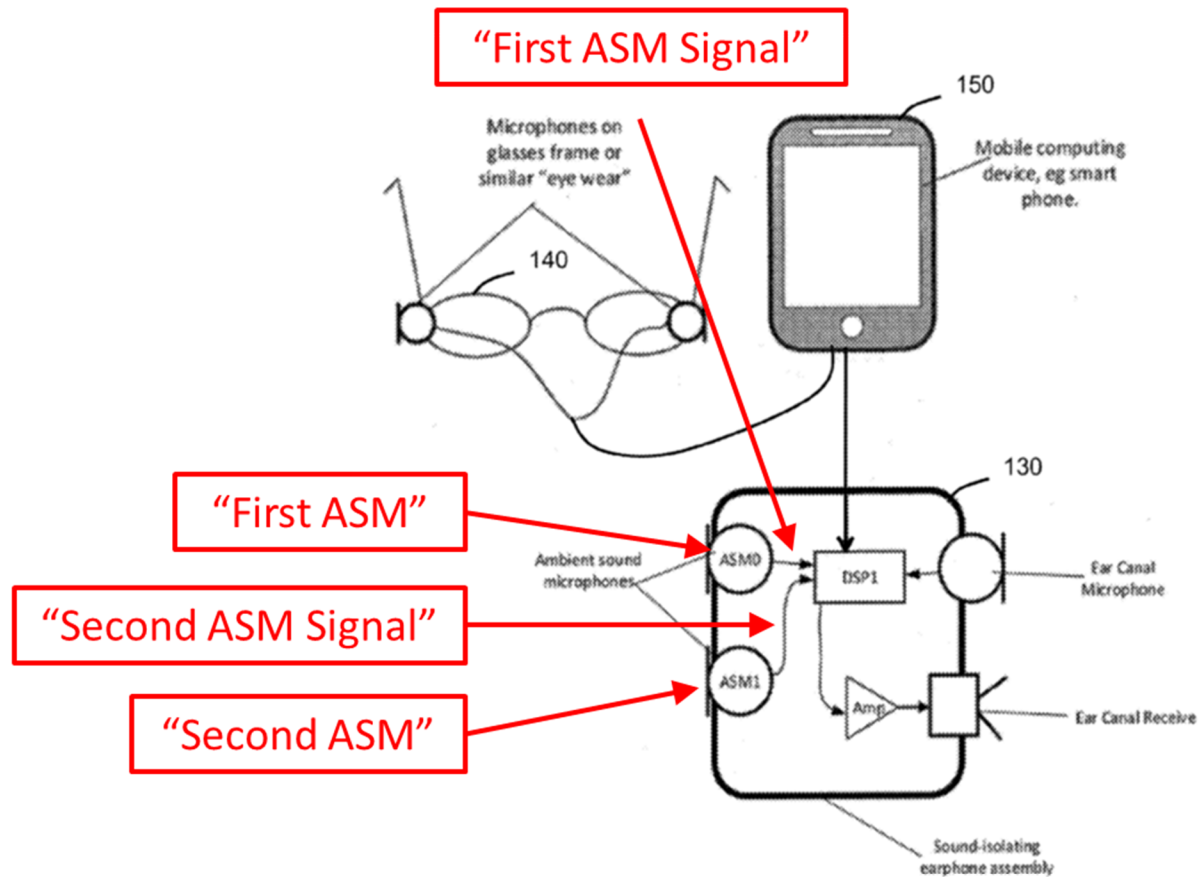
a. [1.pre]¹¹

To the extent the preamble is limiting, *Usher-1* discloses/suggests this feature. *Kyriakakis*, ¶119. *See Usher-1*, ¶[0002].

b. [1.1]

Usher-1 discloses/suggests this feature. *Kyriakakis*, ¶¶120-121. *Usher-1* discloses use of ASMs, e.g., ASM 101/ASM0 and 102/ASM1:

¹¹ The claim language is reproduced in LISTING OF CHALLENGED CLAIMS.



300
FIG. 3A

Usher-I, Fig. 3A. See also Fig. 1A. See also *Id.*, ¶[0021] (“...the earphone 100 includes a processor 103 (e.g., Digital Signal Processor DSP) operatively coupled to one or more ambient sound microphones (101/102)...”). *Usher-I*’s first and second ASMs correspond to the claimed first and second ASMs. *Kyriakakis*, ¶120.

Usher-I further discloses that ASMs 101 (ASM0)/102 (ASM1) output ASM signals, which can be seen in the annotated Fig. 3A, above. *Id.*, ¶[0022] (“The two

ambient sound microphone (101/102) signals...”). The ASM signals correspond to the claimed first and second ASM signals. *Kyriakakis*, ¶121.

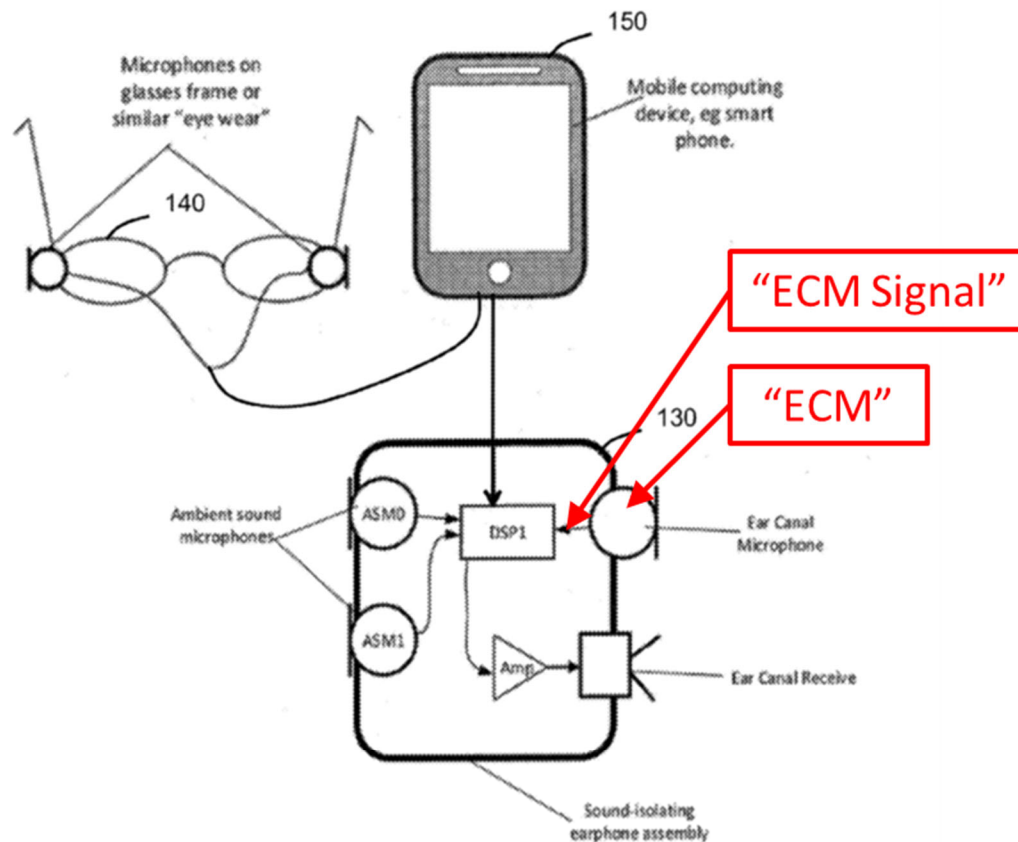
c. [1.2]

See Section IX.A.1.b.

d. [1.3]

Usher-1 discloses/suggests this feature. *Kyriakakis*, ¶¶123-124. *Usher-1*’s

Fig. 3A shows an ECM:



300
FIG. 3A

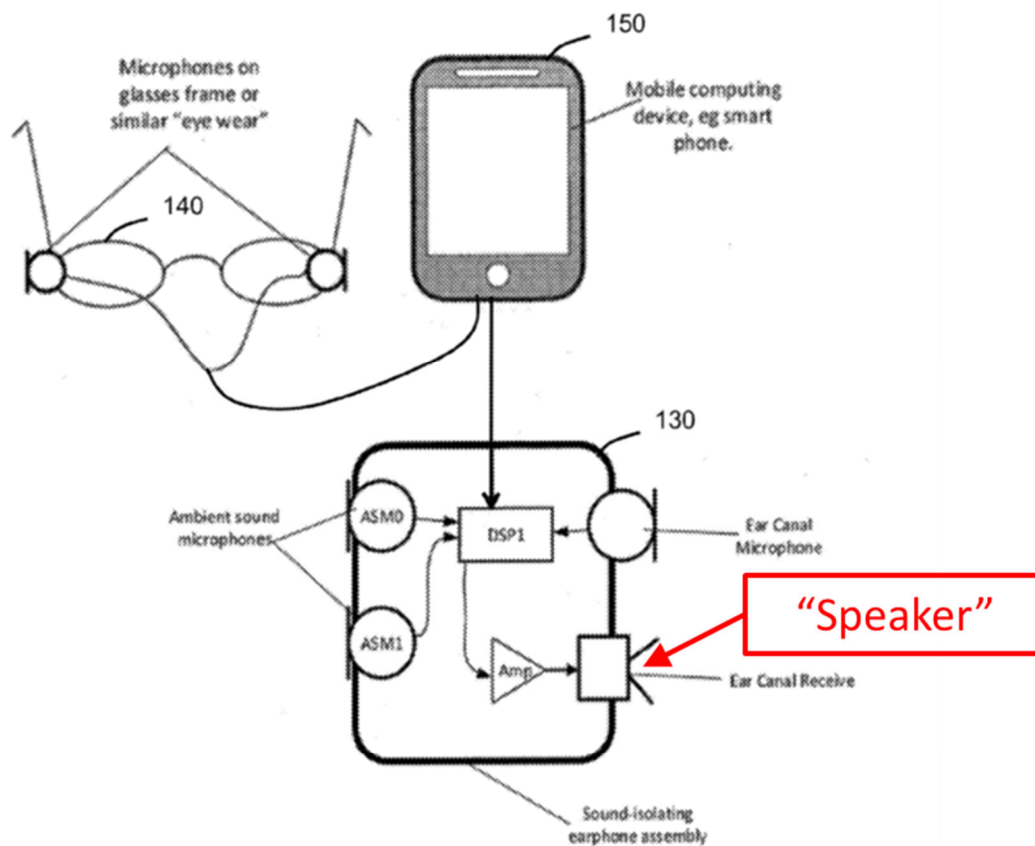
See also *Usher-I*, ¶[0042] (“The system 300 includes the earphone 130 (i.e., earphone 100 equipped with an optional ear canal microphone)...”). See also *Usher-I*’s Fig. 4, element 406 and ¶[0052]. *Usher-I*’s ECM shown in Fig. 3A and ECM 406 correspond to the claimed ECM. *Kyriakakis*, ¶123.

Usher-I’s ECM outputs an ECM signal. *Usher-I*, ¶[0055] (“Microprocessor 416 is operatively coupled to ... ECM 406...”); ¶[0056] (“All transducers shown [in

Fig. 4] can receive or transmit audio signals...”). The signal output by *Usher-1*’s ECM corresponds to the claimed “ECM signal.” *Kyriakakis*, ¶124.

e. [1.4]

Usher-1 discloses/suggests this feature. *Kyriakakis*, ¶¶125-126. *Usher-1* discloses that its earphone has a speaker, which it calls an “ear canal receiver.” See Fig. 1A:



300
FIG. 3A

See also Usher-1, ¶[0021] (“For such purpose, the earphone 100 includes ... an ear canal receiver 105.”); ¶[0022] (“...ear canal receiver (ECR) 105 (i.e., a loudspeaker...”)). *See also* Figs. 3A, 4 (element 414) and ¶[0053].

Usher-1’s ear canal receiver corresponds to the claimed “speaker.” *Kyriakakis*, ¶126.

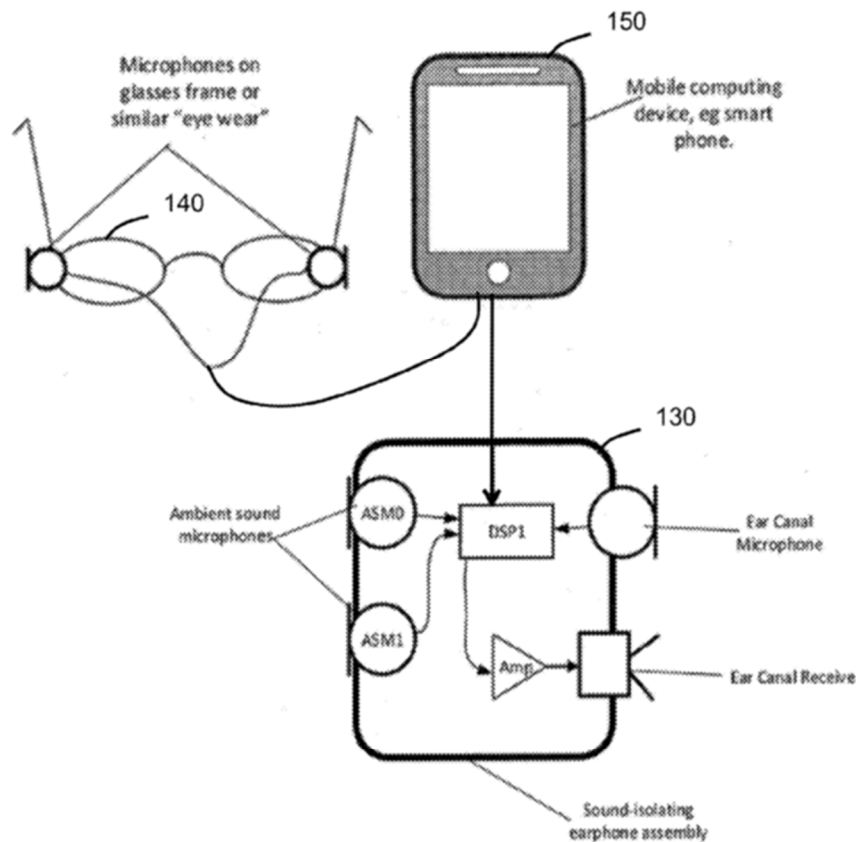
f. [1.5]

Usher-1 discloses/suggests this feature. *Kyriakakis*, ¶¶127-128. *Usher-1* discloses that its earphone has a memory.” *See* Figs. 4 (element 404) and 5 (element 556). *See also Usher-1*, ¶[0055] and [0058]. *Usher-1* discloses that its memory is coupled to a microprocessor. *Usher-1*, ¶[0055] (“Microprocessor 416 is operatively coupled to memory 404...”)). POSITAs would have understood that *Usher-1*’s microprocessor executes instructions to implement *Usher-1*’s method, and that such instructions would be stored in its *memory*. *Kyriakakis*, ¶127.

Thus, *Usher-1*’s memory corresponds to the claimed “memory...” *Kyriakakis*, ¶128.

g. [1.6]

Usher-1 discloses/suggests this feature. *Kyriakakis*, ¶¶129-130. POSITAs would have understood that *Usher-1*’s processor executes instructions to perform operations. *Kyriakakis*, ¶129. *Usher-1* discloses that its processor is coupled to its ASMs, its ECM and its speaker. *See e.g.*, Fig. 3A:



300
FIG. 3A

See also *Usher-1*, Figs. 1A and 4. See also *Id.*, ¶[0021] and [0055].

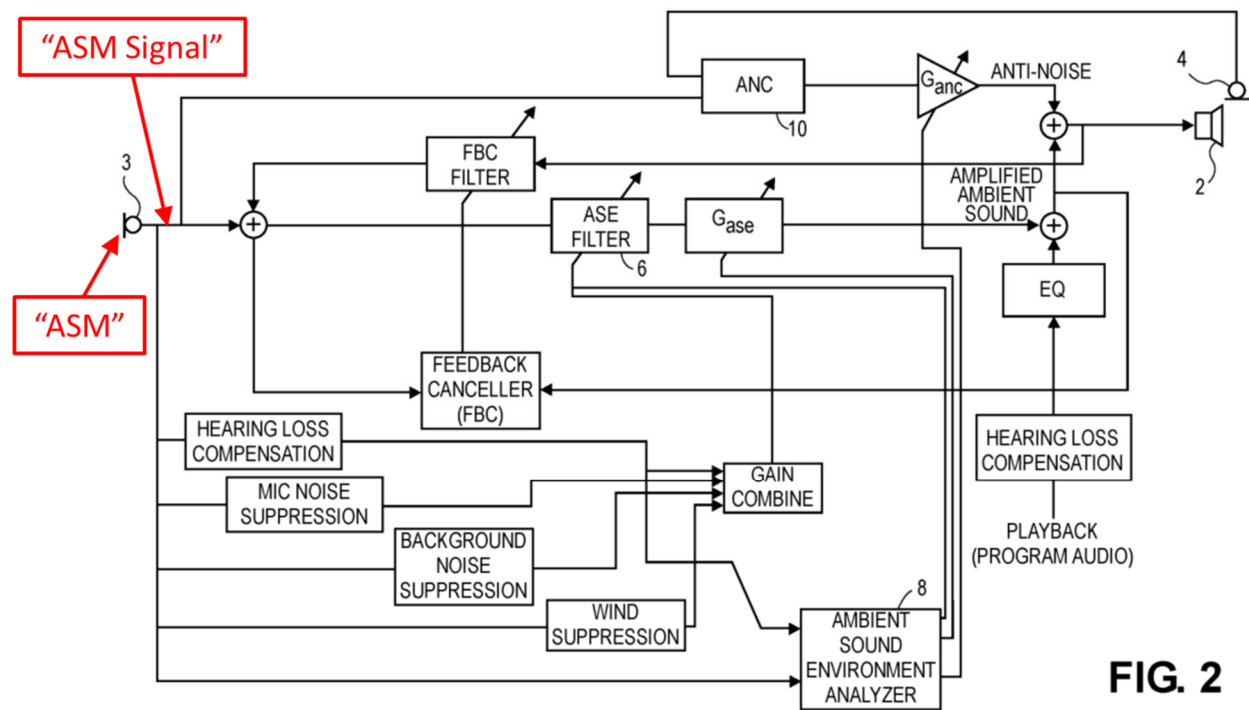
h. [1.7]

Usher-1 in view of *Woodruff-1* discloses/suggests this feature. *Kyriakakis*, ¶131-134. *Usher-1* discloses that its ASMs, e.g., ASMs 101/102, are coupled to its processor. *Usher-1*, ¶[0021] (“...the earphone 100 includes a processor 103 (e.g., Digital Signal Processor DSP) operatively coupled to one or more ambient sound

microphones (101/102)...”).

Usher-1 further discloses that its processor receives the ASM signals from ASMs 101/102. *Id.*, ¶[0022] (“The two ambient sound microphone (101/102) signals are directed to the processor 103...”). POSITAs would have understood that the processor would to execute instructions to perform the operation of receiving these ASM signals. *Kyriakakis*, ¶132. Thus, *Usher-1* discloses this element.

Woodruff-1 also discloses an ASM that outputs an ASM signal. *Woodruff-1* discloses “external microphone 3,” which is shown in Fig. 2:¹²



¹² *Woodruff-1* also discloses use of more than one ASM. See *Woodruff-1*, ¶[0024].

Woodruff-1 further discloses that Fig. 2 operates under the control of a processor. See e.g., *Woodruff-1*, ¶[0022]. Thus, *Woodruff-1* also discloses this element.¹³

i. [1.8]

See Section IX.A.1.h.

j. [1.9]

Usher-1 discloses/suggests this feature. *Kyriakakis*, ¶136-. *Usher-1* discloses that its ECM, e.g., ECM shown in Figs. 1A, 3A and 4, is coupled to its processor. *Usher-1*, ¶[0055] (“...Microprocessor 416 is operatively coupled to ... ECM 406...”). *Usher-1* further discloses that its processor receives an ECM signals from its ECM. *Id.*, ¶[0056] (“All transducers shown can receive or transmit audio signals to a processor 416...”). POSITAs would have understood that for this to happen, the processor would have to execute instructions to perform the operation of receiving these ECM signals. *Kyriakakis*, ¶136. Thus, *Usher-1* discloses this element.

k. [1.10]

Usher-1 in view of *Woodruff-1*/*Woodruff-2* discloses/suggests this feature. *Kyriakakis*, ¶¶137-148. As discussed, *Usher-1* discloses that the its ASMs output

¹³ The reasons why POSITAs would have been motivated to combine *Woodruff-1* with *Usher-1* will be discussed in Section IX.A.1.k(i).

ASM signals. *See* Sections IX.A.1.b and IX.A.1.c. As also discussed, *Usher-1* discloses that its ASM signals are sent to its processor. *See* Sections IX.A.1.h and IX.A.1.i. *Usher-1*, however, does not explicitly teach generating a new signal, i.e., “a first signal,” from one or both of the first and second ASM signals. POSITAs would have recognized that *Usher-1* in view of *Woodruff-1*/*Woodruff-2* discloses the generation of such a “first signal.” *Kyriakakis*, ¶137.

Woodruff-1 is very similar to *Usher-1* in that both disclose what each refer to as a “pass-through” mode. *Usher-1*, ¶[0021], [0026], [0029] and [0033]-0034]; *Woodruff-1*, ¶[0045]. *Woodruff-1* discloses “an audio system that has an ambient sound enhancement function, also referred to here as a transparency mode of operation, in which an against-the-ear audio device having one or more speakers converts a digitally processed version of an input audio signal into sound.” *Woodruff-1*, ¶[0004]. *Woodruff-1* discloses that its transparency mode is a “pass-through” mode (*Id.*, ¶[0045]), which POSITAs would have understood has the same purposes as *Usher-1*’s “ambient sound pass-through to enhance situation awareness” embodiment (*Usher-1*, ¶[0029]). *Kyriakakis*, ¶139. *Woodruff-1*’s signal processing for implementing its pass-through mode is disclosed in Fig. 2:

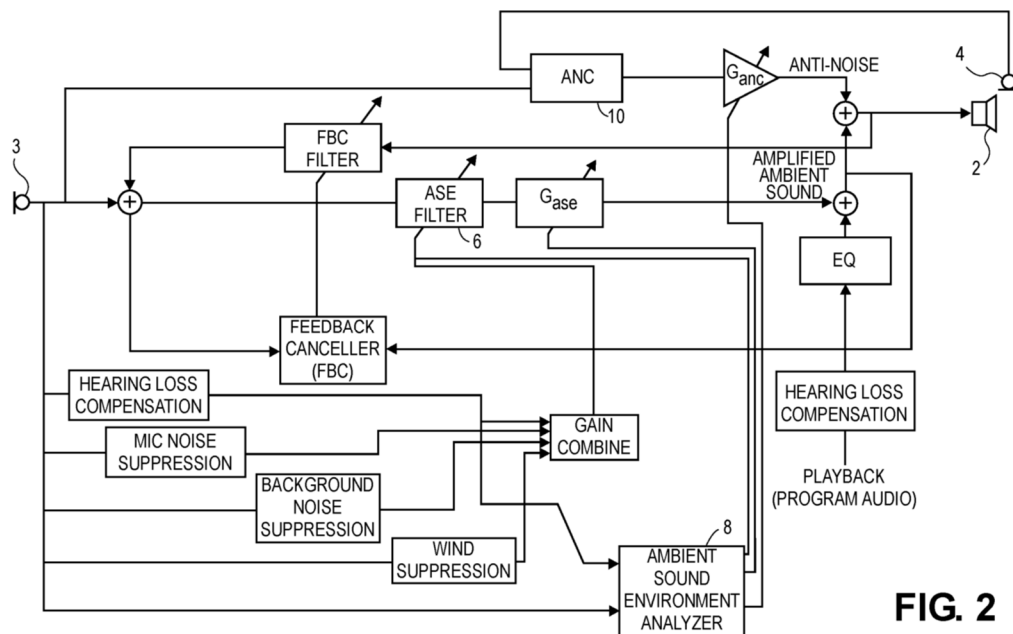


FIG. 2

Woodruff-1's Fig. 2 is "a block diagram of an audio system that combines personalized ambient sound enhancement and acoustic noise cancellation for a user of the against the ear device 1." *Woodruff-1*, ¶[0022]. *Woodruff-1* discloses that "[a]mbient sound in an acoustic environment of a user who is wearing (using) the against the ear audio device 1 is picked up by the external microphone 3." *Id.* POSITAs would have understood that *Woodruff-1*'s external microphone 3 is an ASM, and corresponds to at least one of the claimed first and second "ambient sound microphone(s), and further serves the same purpose as *Usher-1*'s ambient sound microphones 101/102. *Kyriakakis*, ¶140. Likewise, POSITAs would have understood that the output of *Woodruff-1*'s microphone 3 corresponds to at least one of the claimed first or second "ASM signals." *Id.* *Woodruff-1*'s Fig. 2 discloses feedback cancellation components (e.g., FBC filter and Feedback Canceller), but

does not describe them in detail.

As discussed in Section V.C, *Woodruff-2* contains essentially the same feedback cancellation components, which it does describe. In particular, *Woodruff-2*, like *Woodruff-1*, discloses use of a Feedback Canceller and an FBC Filter. As is seen, the output of ASE Filter 6 (discussed in Section IX.A.1.1), has a gain applied to it (shown as amplifier A in *Woodruff-2*'s Fig. 3). The output of that amplified signal is input to FBC Filter 10.

Woodruff-1 is similar, although it has additional signal processing. Specifically, the output of *Woodruff-1*'s ASE Filter 6 also has a gain applied (see gain block G_{ase}). Rather than inputting this signal directly into FBC Filter, as in *Woodruff-2*, *Woodruff-1* first combines the this signal with the program audio (e.g., music). *Woodruff-1*, ¶[0025] (“The ambient sound as picked up in a sound pickup channel (e.g., a microphone signal from the external microphone 3) is filtered in a frequency-dependent transparency gain block G_{ase} , which is controlled by an ambient sound environment analyzer 8 according to the hearing profile of the user (based on input from the hearing loss compensation block.) The ambient sound content is then combined with a playback signal which contains program audio. The program audio may be music, the voice of a far end user during a phone call, or the voice of a virtual assistant program, which has not yet been converted to sound in the ambient environment.”). This combined signal is combined with an anti-noise

signal. *Woodruff-1*, ¶[0029] (“The anti-noise signal is combined with the ambient sound signal before they together drive the speaker 2 of the against the ear audio device.”). As seen in Fig. 2, this signal is also input to FBC Filter 10. *Kyriakakis*, ¶142.

Woodruff-2 further discloses that “[t]he output of the FBC filter 10 may be added to input audio signal coming from the external microphone 3 to result in a combined signal at the input of the ASE filter 6.”¹⁴ *Id.*, 7:33-36. The same feature is shown in *Woodruff-1*’s Fig. 2. POSITAs would have recognized that the adder shown *Woodruff-1*/*Woodruff-2* adds the output of FBC filter 10 to the signal output by ASM 3 (i.e., a claimed ASM signal), and that this adder generates a new signal, which corresponds to the recited “first signal.” *Kyriakakis*, ¶143.

Woodruff-2’s Fig. 3 has been annotated to show that microphone 3 corresponds to a claimed ASM, that its output corresponds to a claimed ASM signals, and that the output of the adder corresponds to the claimed “first signal.”

¹⁴ POSITAs would have understood that when *Woodruff-2* uses the abbreviation “ASE” as shorthand for “ambient sound enhancement.” *Woodruff-2*, 1:39-43. See also *Kyriakakis*, Footnote 8. Given the overlapping inventors and the fact that *Woodruff-1* and *Woodruff-2* have the same assignee and similar drawings, POSITAs would have understood that “ASE” has the same meaning in both.

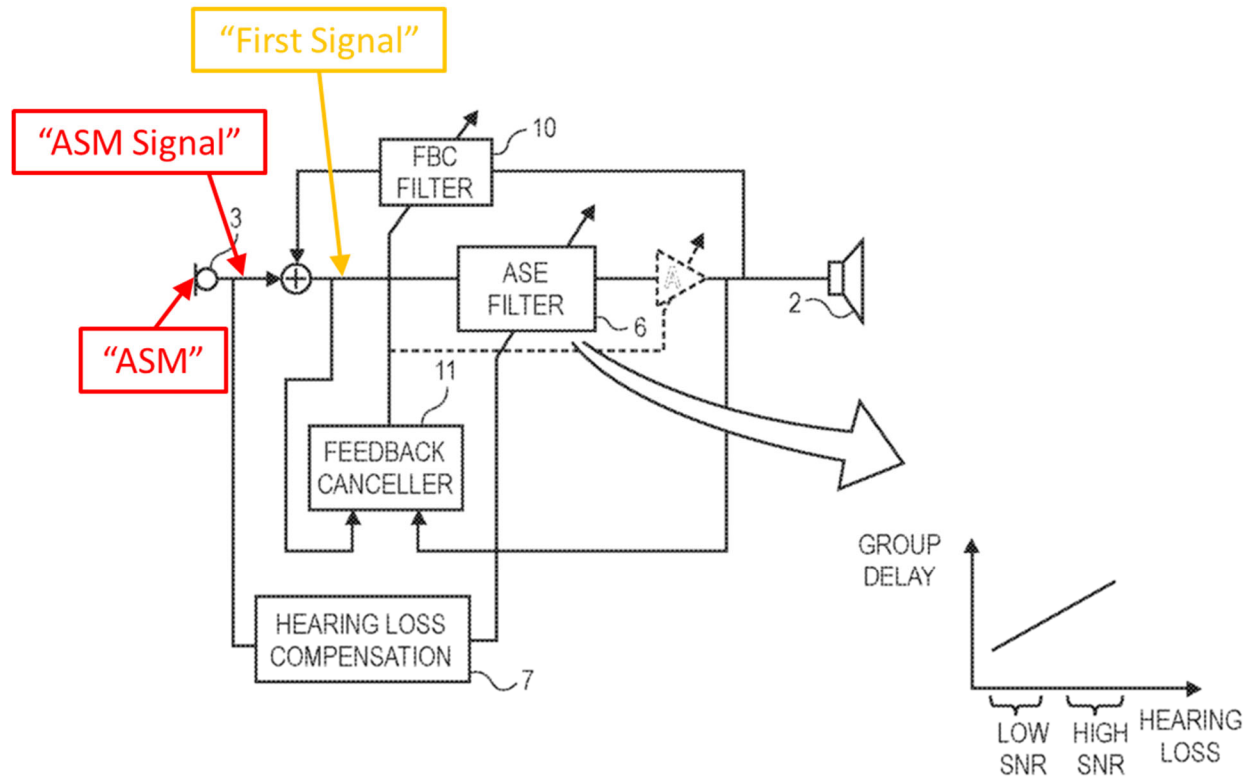
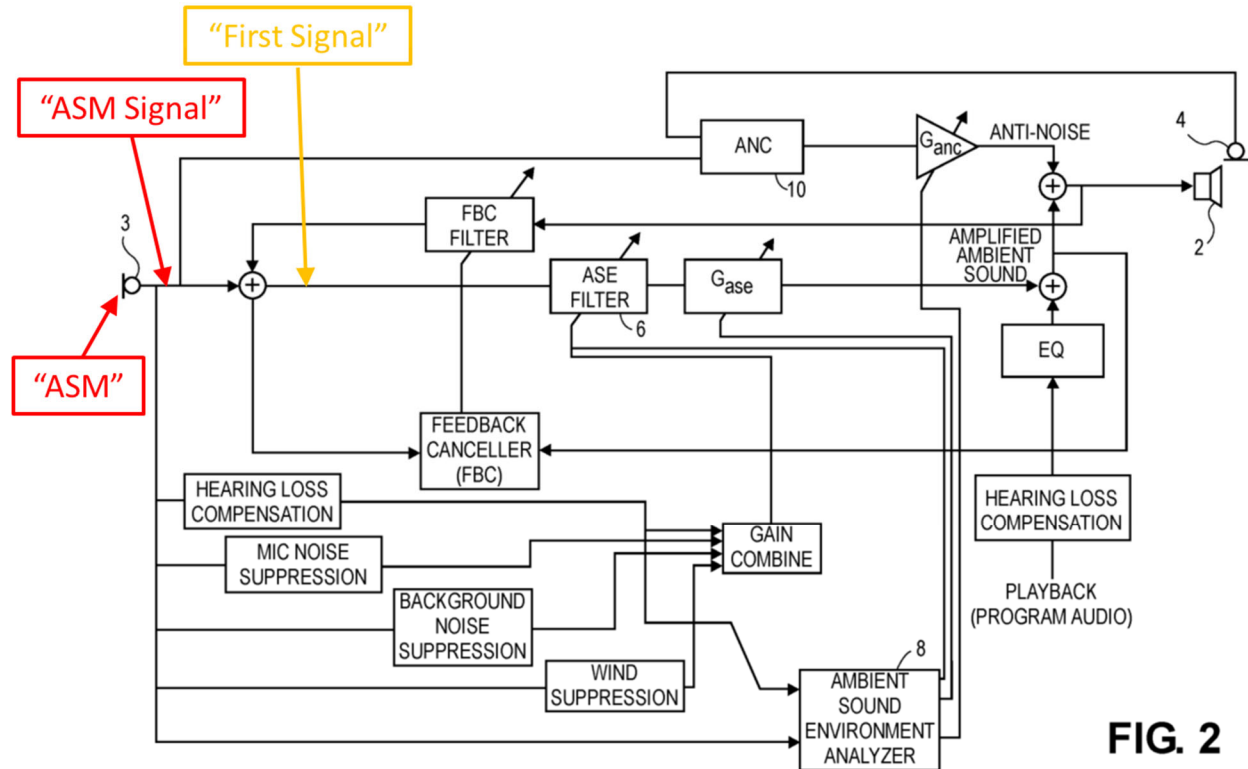


FIG. 3

It is also shown in *Woodruff-1's* annotated Fig. 2:



Kyriakakis, ¶144.

Note that claim 1 does not specify how the “first signal” is generated or what form it takes. The language only requires that the “first signal” be generated from one or both of the recited first or second “ASM signals.” Of course, given that the claim only requires that one of the ASM signals be used, the fact that *Woodruff-1/Woodruff-2* only explicitly disclose a first ASM microphone and hence only one ASM signal is irrelevant. *Id.*, ¶145.

(i) POSITAs Would Have Been Motivated To Combine *Woodruff-1* and *Woodruff-2* With *Usher-1*

Usher-1, *Woodruff-1*, and *Woodruff-1* are each directed to earphone devices.

Usher-1, ¶[0001], *Woodruff-1*, ¶[0002], *Woodruff-2*, 1:11-15. They are structurally similar, since, as discussed in Sections **Error! Reference source not found.-V.C**, all three have an ASM, an ECM and in-ear speaker. *Kyriakakis*, ¶146. Moreover, both *Usher-1* and *Woodruff-1* describe “pass-through” modes, in which the earphone can reproduce audio from the ambient environment in the user’s ear. *Usher-1*, ¶[0021], [0026], [0029] and [0033]-0034]; *Woodruff-1*, ¶[0045].

POSITAs would have been motivated to implement *Woodruff-1*’s signal processing for implementing such a “pass-through” mode with *Usher-1*’s earphone. First, POSITAs would have recognized that *Usher-1* and *Woodruff-1* are in the same field (earphones), and would have been motivated to use *Woodruff-1*’s signal processing teachings for noise suppression. For example, a person of ordinary skill would have recognized that *Woodruff-1*’s teachings on the suppression of microphone noise, background noise, wind noise are all needed elements to improve the performance of *Usher-1*’s earphone.. *Kyriakakis*, ¶147.

POSITAs would have recognized that *Woodruff-2* explains certain features of *Woodruff-1*, and that *Woodruff-2* provides further motivation for incorporating *Woodruff-1* (and hence *Woodruff-2*) into *Usher-1*. For example, *Woodruff-2* teaches that its FBC Filter attempts to remove acoustic coupling of its speaker into the ASM. *Woodruff-2*, 7:30-36. *Woodruff-2* teaches that this is particularly important when feedforward gain is applied to the ambient audio signal. *Id.* *Woodruff-2* further

teaches that its FBC Filter also helps when FBC control loop becomes unstable. *Id.*, 7:37-52. POSITAs would have understood that these conditions could result in what is known as “howling,” an unstable condition that earphone users would find at best annoying and at worst very uncomfortable, meaning that POSITAs would have been motivated to reduce the possibility of it occurring. *Kyriakakis*, ¶148.

I. [1.11]

Usher-1 in view of *Woodruff-1* and *Woodruff-2* discloses/suggests this feature. *Kyriakakis*, ¶¶149-152. *Usher-1* does not explicitly state that it generates an “ambient sound signal” from either the “first signal or a modified first signal.” However, both *Woodruff-1* and *Woodruff-1* disclose this. As discussed, both *Woodruff-1* and *Woodruff-2* disclose a signal comprised of the combination of the output from ASM microphone 3 and FBC Filter, which corresponds to the claimed “first signal.” See Section IX.A.1.k. Both *Woodruff-1* and *Woodruff-2* also disclose that this “first signal” is input to ASE Filter 6. *Woodruff-2*, 4:63-65, 5:17-29. POSITAs would have understood that the output of *Woodruff-1*/*Woodruff-2*’s ASE Filter 6 corresponds to the claimed “ambient sound signal.”

Woodruff-2’s Fig. 3 has been annotated to show that the output of ASE Filter 6 corresponds to the claimed “ambient sound signal:”

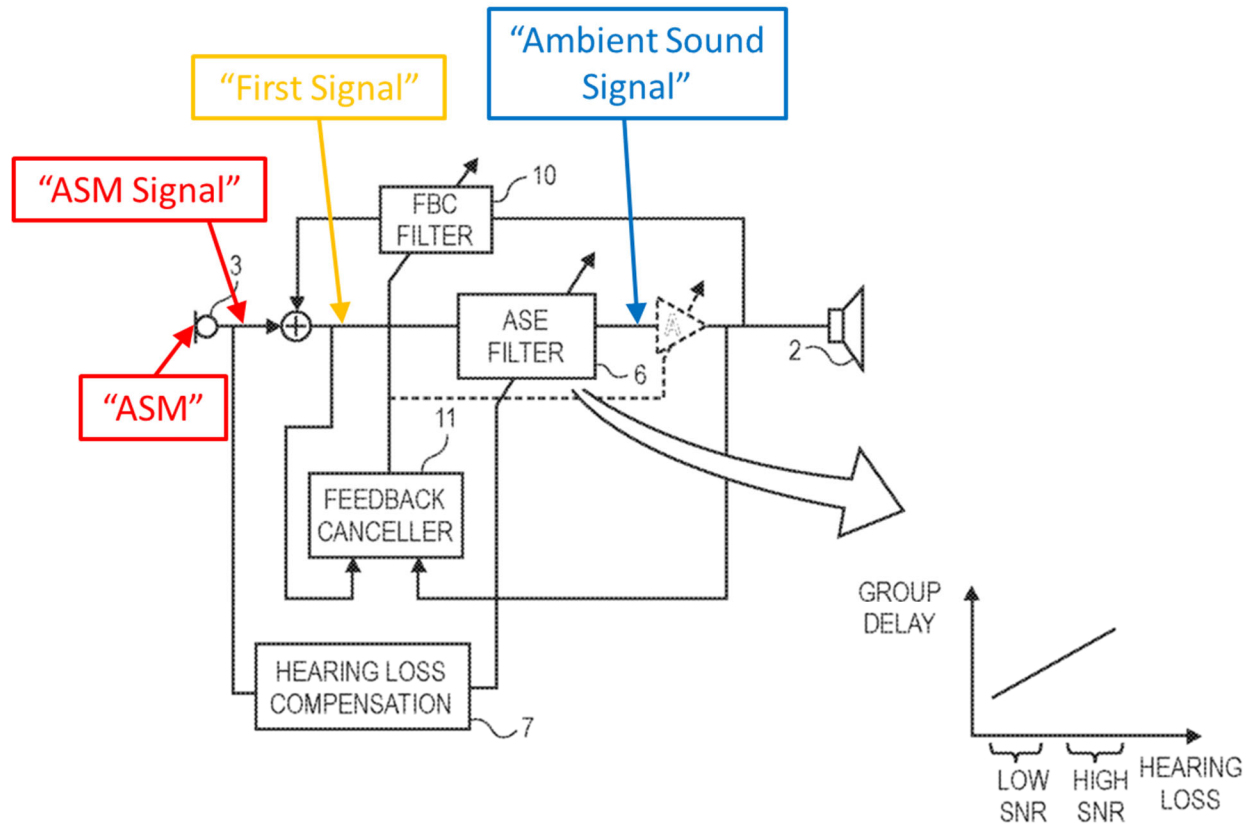


FIG. 3

See also annotated Woodruff-1's Fig. 2:



Note that claim 1 does not specify how the “ambient sound signal” is generated or what form it takes. All that is required is that the recited signal be generated from “the first signal or a modified first signal,” and that it relate to “ambient sound.” *Kyriakakis*, ¶151.

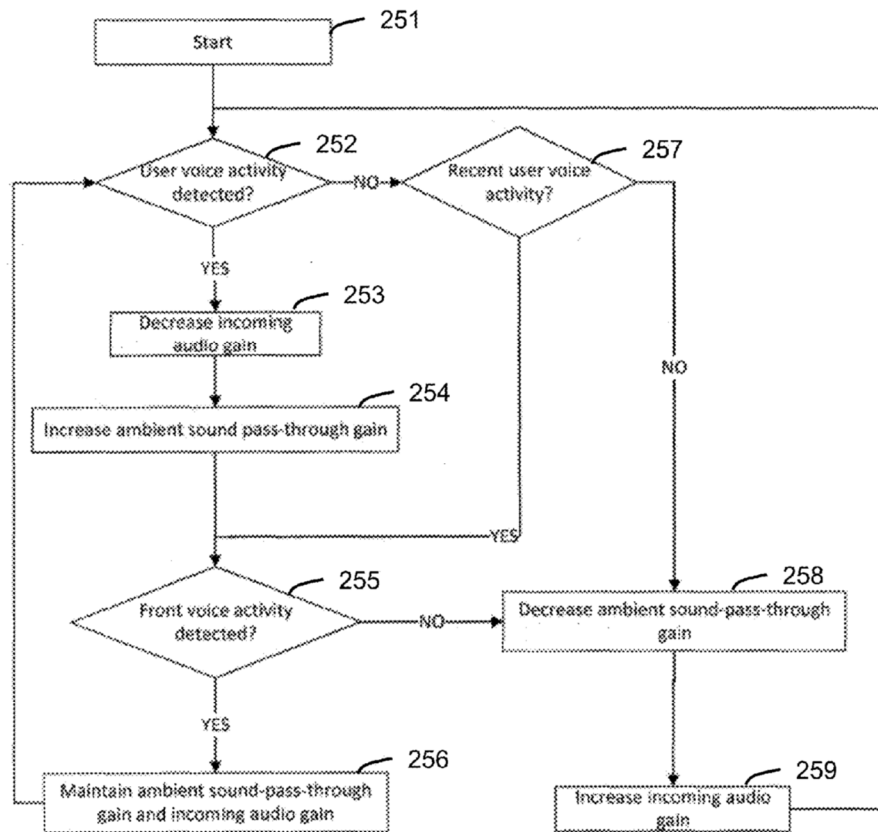
Woodruff-1 discloses that the output of ASE Filter 6 is an “ambient” sound signal. *Woodruff-1* notes that ambient sound is captured by ASM 3 and then provided to ASE Filter 6. *Woodruff-1* further discloses that ASE Filter 6 shapes the ambient sound for output. *Woodruff-1*, ¶[0022] (“The ASE filter 6 may be composed of a cascade of digital filters that spectrally shape the ambient sound pickup channel for purposes of different types of noise suppression, e.g., microphone noise,

background noise, and wind.”). This signal is applied to gain block G_{ase} , which amplifies the signal for playback by speaker 2. Indeed, Fig. 2 refers to this signal as “AMPLIED AMBIENT SOUND.” Thus, the output of ASE Filter 6 is plainly an “ambient sound signal,” just as claimed. *Kyriakakis*, ¶152.

m. [1.12]

Usher-1 in view of *Woodruff-1* and *Woodruff-2* discloses/suggests this feature. *Kyriakakis*, ¶¶153-154. This element requires determining if the user is speaking by using one of any number of signals.¹⁵ *Usher-1*’s Fig. 2B “illustrates a flowchart of a method 250 for controlling an ambient sound pass-through to enhance situation awareness in accordance with an exemplary embodiment.”

¹⁵ The list separates the recited signals with the word “or,” which means that if any one of the signals is in the prior art, the element is disclosed. *See e.g., Schumer v. Lab. Computer Sys., Inc.*, 308 F.3d 1304, 1311-12 (Fed. Cir. 2002).



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FIG. 2B

Usher-1, ¶[0029]. *Usher-1* further discloses that “[a]t step 252, the processor 103 monitors sound from the ambient sound microphone(s) 101/102 and determines when user voice activity is detected. The processor 103 performs voice activity detection (VAD) to analyze voice characteristics and properties and distinguish between the user’s spoken voice and environmental ambient sounds.” *Id.*, at ¶[0030].

Thus, *Usher-1* discloses that its device uses signals from its ASMs, (which correspond to the claimed “ASM signal(s)” to perform user voice activity detection.

Since the claim only requires detecting if the user is speaking by analyzing the first or second “ASM signal,” POSITAs would have recognized *Usher-1* in view of *Woodruff-1/Woodruff-2* discloses this element. *Kyriakakis*, ¶¶153-154.

n. [1.13]

Usher-1 in view of *Woodruff-1/Woodruff-2* discloses/suggests this feature. *Kyriakakis*, ¶¶155-160. This element requires adjusting “at least one” of a number of recited signals when it is detected that the user is speaking.” One of the recited signals is the “audio content signal.” *Usher-1* discloses that “[i]f at step 252 user voice activity is detected, the gain of the incoming audio signal is decreased at step 253....” *Usher-1*, ¶[0030].

Usher-1 discloses that the “[t]he processor 103 can also receive an incoming audio signal 107, e.g. a voice or music signal from a mobile device.” *Id.*, ¶[0022]. As discussed in Section VIII.A, the ‘965*Pat*’s “audio content signal” is a signal containing audio content. Thus, POSITAs would understand that the “incoming audio signal” described in *Usher-1* corresponds to the claimed “audio content signal.” *Kyriakakis*, ¶156. Because *Usher-1* discloses that gain is decreased when “user voice activity is detected,” *Usher-1* teaches that its “audio content signal” is “adjusted ... when it is detected that the user is speaking.”

This element also recites that the “ambient sound signal” is one of the signals that can be adjusted when user speech is detected. *Usher-1* discloses that “[i]f at

step 252 user voice activity is detected, ...the ambient sound pass-through gain is increased at step 254.” *Usher-1*, ¶[0030]. *See also id.*, ¶[0040] (“[The processor] determines a direction and proximity of the individual with respect to the user wearing the earphone (*see id.*, FIGS. 2A and 2C), and adjusts the gain (185/187) of the ambient sound pass-through of the ambient sound microphone 101 as a function of the direction in accordance with the previous methods (see FIGS. 1B and 2B).”), ¶[0041] (“The step of adjusting the ambient sound pass-through includes increasing the gain of the ambient sound from the ambient sound microphone delivered by the internal speaker when voice activity is detected above a threshold, and/or decreasing the gain of the ambient sound from the ambient sound microphone delivered by the internal speaker when voice activity is detected below a threshold.”).

Note that claim 1 does not specify who the recited “user” is. If the recited “user” refers to the person wearing the earphone, *Usher-1* plainly discloses this, as discussed. If the recited “user” can be either the wearer or a person speaking to the wearer. In this case, *Woodruff-1* discloses this, as discussed in the immediately preceding paragraph.

As discussed in Section IX.A.1.k, when *Usher-1* is combined with *Woodruff-1/Woodruff-2*, the ASM signal produced by ASM microphone 101/102 is combined with the output of *Woodruff-1/Woodruff-2*’s FBC Filter 10 to produce the claimed “first signal,” which in turn is filtered by ASE Filter 6 to produce the claimed

“ambient sound signal.” *Kyriakakis*, ¶159. *Woodruff-1* is similar to *Usher-1*, as it also discloses that the signal output by its ASE Filter 6, corresponding to the claimed “ambient sound signal” can be adjusted when the user speaks. *Woodruff-1*, ¶[0042] (“But when the wearer starts talking to someone at the concert, the determined user context changes, e.g., ‘wearer is talking with someone in their environment’, because the system detects speech by another person in the environment and/or *speech by the wearer*, the G_{ase} block is signaled to emphasize speech frequencies (e.g., by appropriately shaping the filtering function of the Gase block.)’”); ¶[0043] (“In yet another aspect of operating in transparency mode, the system reduces the transparency gain that it applies (in the Gase block), differently in different user contexts. For instance, when the determined context indicates that the wearer is ‘talking over’ ambient noise, the transparency gain is reduced more than when the wearer is passively listening to their environment.”); ¶[0044] (“...the wearer may be at a concert when during a lull in the performance the wearer starts talking to a friend next to them (who is also at the concert.) In that case, the audio system detects a drop in the ambient noise/sound level (below a threshold) and perhaps also a contemporaneous increase in speech activity. The speech activity may be either the wearer’s own voice or that of the friend nearby. In response the system may reduce its anti-noise output (e.g., raise attenuation by G_{anc}), and perhaps increase its transparency gain G_{ase} that it applies to an ambient sound pickup channel, and also

perhaps emphasize the speech frequency range.”).

Thus, POSITAs would have recognized that both *Usher-1* and *Woodruff-1/Woodruff-2* disclose adjusting the sound from the ambient environment when it is detected that the user is speaking. *Kyriakakis*, ¶¶159-160. Thus, claim 1 is obvious.

2. Claim 2

The *Usher-1-Woodruff-1/Woodruff-2* combination discloses/suggests this claim. *Kyriakakis*, ¶¶161-162. As discussed in Section IX.A.1.n, *Usher-1* discloses adjusting the “audio content signal” when it is detected that the user is speaking. And while POSITAs would have understood that both *Usher-1* and *Woodruff-1* will reduce the volume of the “audio content signal” when it is detected that the user is speaking, *Usher-1* explicitly says this is the case. *See Usher-1*, ¶[0030] (“If at step 252 user voice activity is detected, the gain of the incoming audio signal is decreased at step 253 and the ambient sound pass-through gain is increased at step 254.”), ¶[0041] (“The step of adjusting the mixing gain of an audio content signal includes decreasing a volume of the audio content signal delivered to the internal speaker when voice activity is detected above a threshold...”)). Thus, claim 2 is obvious.

3. Claim 3

The *Usher-1-Woodruff-1/Woodruff-2* combination discloses/suggests this claim. *Kyriakakis*, ¶¶163-165. *Usher-1* discloses that when it is detected that the user is speaking, the ambient sound detected by its ASM microphone(s) is amplified.

See Usher-1, ¶[0030] (“If at step 252 user voice activity is detected, the gain of the incoming audio signal is decreased at step 253 and the ambient sound pass-through gain is increased at step 254.”),¹⁶ ¶[0041] (“The step of adjusting the ambient sound pass-through includes increasing the gain of the ambient sound from the ambient sound microphone delivered by the internal speaker *when voice activity is detected* above a threshold”). POSITAs would have understood that increasing the gain of the ambient sound would amplify the ambient sound. *Kyriakakis*, ¶164.

As discussed in Sections IX.A.1.1 and IX.A.1.n, in the *Usher-1-Woodruff-1/Woodruff-2* combination, the output of *Woodruff-1*’s ASE Filter 6 is the recited “ambient sound signal.” And *Woodruff-1*, just like *Usher-1*, discloses that the signal output from ASE Filter 6 can be amplified when the user is speaking. *See Woodruff-1*, ¶[0044] (“...the wearer may be at a concert when during a lull in the performance the wearer starts talking to a friend next to them (who is also at the concert.) In that case, the audio system detects a drop in the ambient noise/sound level (below a threshold) and perhaps also a contemporaneous increase in speech activity. The speech activity may be either the wearer’s own voice or that of the friend nearby. In

¹⁶ Because claim 3 depends from claim 2, claim 3 requires both that the audio content signal have its volume reduced and that the ambient sound signal be amplified. *Usher-1* describes this. *See Usher-1*, ¶[0030]

response the system may ... increase its transparency gain Gase that it applies to an ambient sound pickup channel.”). Thus claim 3 is obvious. *Kyriakakis*, ¶165.

4. Claim 4

The *Usher-1-Woodruff-1/Woodruff-2* combination discloses/suggests this claim. *Kyriakakis*, ¶¶166-168. Claim 4 recites limitations similar to claim 3, with the difference being that claim 4 is not dependent on claim 2, and claim 4 recites increasing the volume of the ambient sound signal. As discussed in Section IX.A.3, *Usher-1* discloses increasing the gain of the signal output by its ASM microphones 101/102. *Usher-1*, ¶[0030], [0041]. POSITAs would have understood that “increasing the gain” would result in increasing the volume of the signal having the gain applied thereto. *Kyriakakis*, ¶166.

As also discussed in Section IX.A.3, *Woodruff-1* also discloses increasing gain of the signal output from ASE Filter 6 (the recited “ambient audio signal”) when the user is speaking. See *Woodruff-1*, ¶[0044] (“...the wearer may be at a concert when during a lull in the performance the wearer starts talking to a friend next to them (who is also at the concert.) In that case, the audio system detects a drop in the ambient noise/sound level (below a threshold) and perhaps also a contemporaneous increase in speech activity. The speech activity may be either the wearer's own voice or that of the friend nearby. In response the system may ... increase its transparency gain Gase that it applies to an ambient sound pickup channel.”).

Since increasing the gain of this signal will increase its volume, POSITAs would have understood that claim 4 is obvious. *Kyriakakis*, ¶168.

5. Claim 6

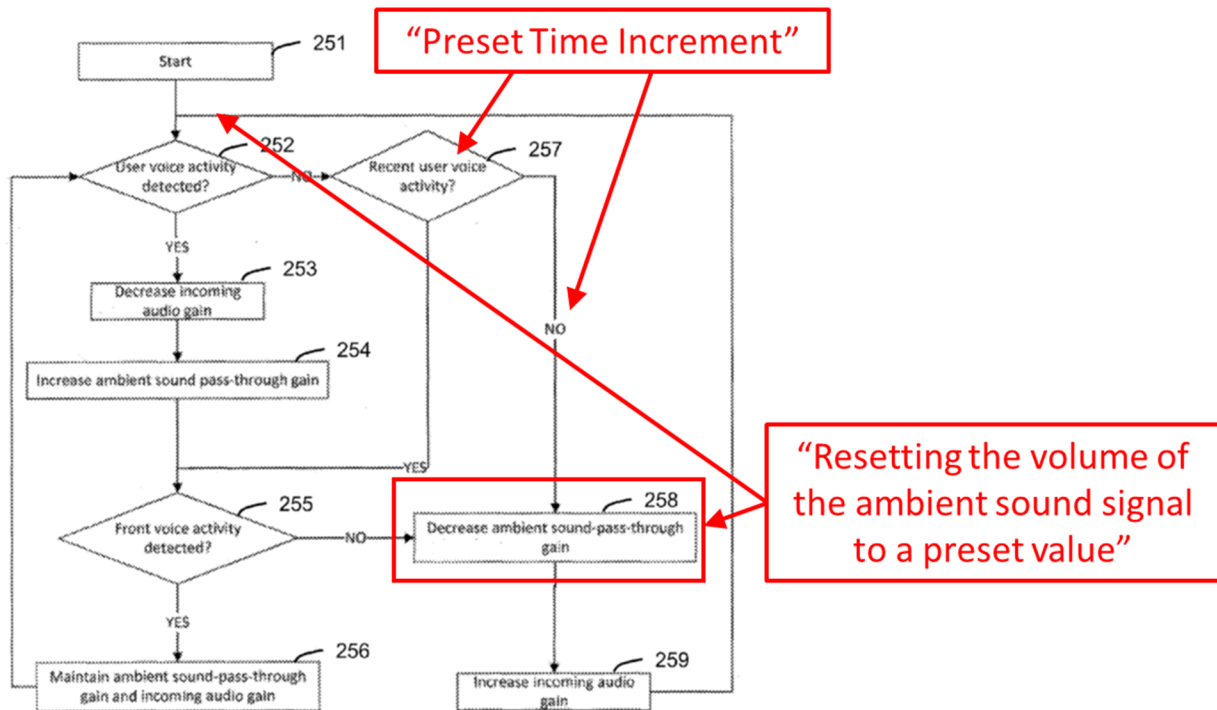
The *Usher-1-Woodruff-1/Woodruff-2* combination discloses/suggests this claim. *Kyriakakis*, ¶¶169-173. *Usher-1* discloses that “[w]hen the user ceases to speak, or if no voice activity is detected, the voice activity timer is referenced to determine if there was any recent user voice activity at step 257.” *Usher-1*, ¶[0033]. *Usher-1* further discloses “two timer scenarios.” *Id.* The first is where the voice activity timer is below a “determined threshold,” which *Usher-1* states can be ten seconds. *Id.* The second is where “user voice activity is not detected at step 252 but the voice activity timer is above the determined threshold.” *Id.* In this scenario, “...the gain of the ambient sound pass-through gain is decreased at step 258...” *Id.*

POSITAs would have understood that *Usher-1*’s “determined threshold” corresponds to the claimed “preset time increment,” since, as *Usher-1* discloses, it is used to determine whether “there was any recent user voice activity.” *Id.* See also *Kyriakakis*, ¶170.

Usher-1 discloses that when the “determined threshold” has been exceeded, “the gain of the ambient sound pass-through gain is decreased at step 258...” *Usher-1*, ¶[0033]. According to *Usher-1*, once this gain has been applied, “the method then continues back to step 252 to monitor for user voice activity.” *Id.* POSITAs would

have understood that this means that the gains have been set to a “preset value” since the method reverts back to its original state of monitoring for voice activity.

Kyriakakis, ¶172. This process is seen in *Usher-1*’s Fig. 2B (annotated):



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FIG. 2B

Finally, as discussed in Sections IX.A.1.1 and IX.A.1.n, in the *Usher-1-Woodruff-1/Woodruff-2* combination, the output of *Woodruff-1*’s ASE Filter 6 is recited “ambient sound signal,” and this is the signal that is “reset,” as described in *Usher-1*. *Kyriakakis*, ¶173.

6. Claim 7

See Section IX.A.5.

7. Claim 8

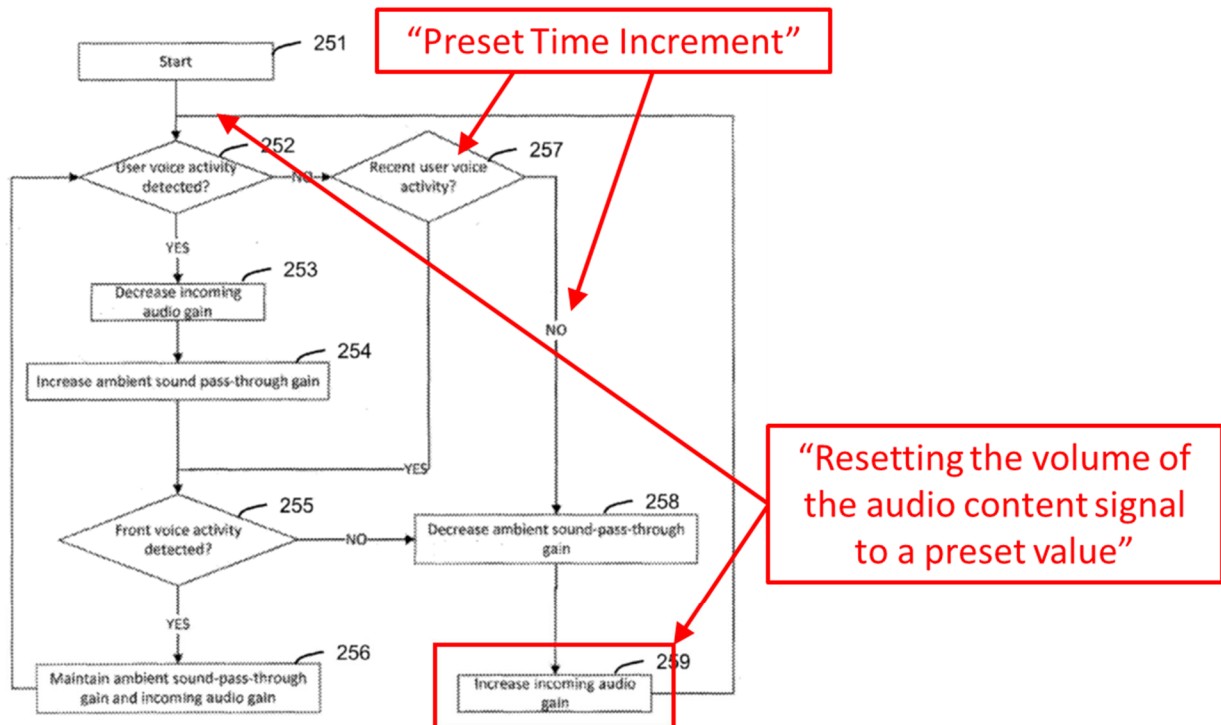
The *Usher-1-Woodruff-1/Woodruff-2* combination discloses/suggests this claim. *Kyriakakis*, ¶¶175-178. Claim 8 is similar to claim 6, with the only difference being that claim 8 requires that the volume of the “audio content signal” be reset. *Usher-1* discloses that “[w]hen the user ceases to speak, or if no voice activity is detected, the voice activity timer is referenced to determine if there was any recent user voice activity at step 257.” *Usher-1*, ¶[0033]. As discussed in Section IX.A.5, *Usher-1* further discloses two timer scenarios, one of which is where “user voice activity is not detected at step 252 but the voice activity timer is above the determined threshold.” *Id.* In this scenario, “...the gain of the incoming audio signal is increased at step 259.” *Id.*

POSITAs would have understood that that *Usher-1*’s “determined threshold” corresponds to the claimed “preset time increment,” since, as *Usher-1* discloses, it is used to determine whether “there was any recent user voice activity.” *Id.* See also *Kyriakakis*, ¶176.

Usher-1 discloses that when the “determined threshold” has been exceeded, “the gain of the ambient sound pass-through gain is decreased at step 258, and the gain of the incoming audio signal is increased at step 259.” *Usher-1*, ¶[0033]. According to *Usher-1*, once these gains have been applied, “the method then continues back to step 252 to monitor for user voice activity.” *Id.* POSITAs would

have understood that this means that the gains have been set to a “preset value” since the method reverts back to its original state of monitoring for voice activity.

Kyriakakis, ¶177. This process is seen in *Usher-1*’s Fig. 2B (annotated):



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FIG. 2B

Thus, claim 8 is obvious. *Kyriakakis*, ¶178.

8. Claim 9

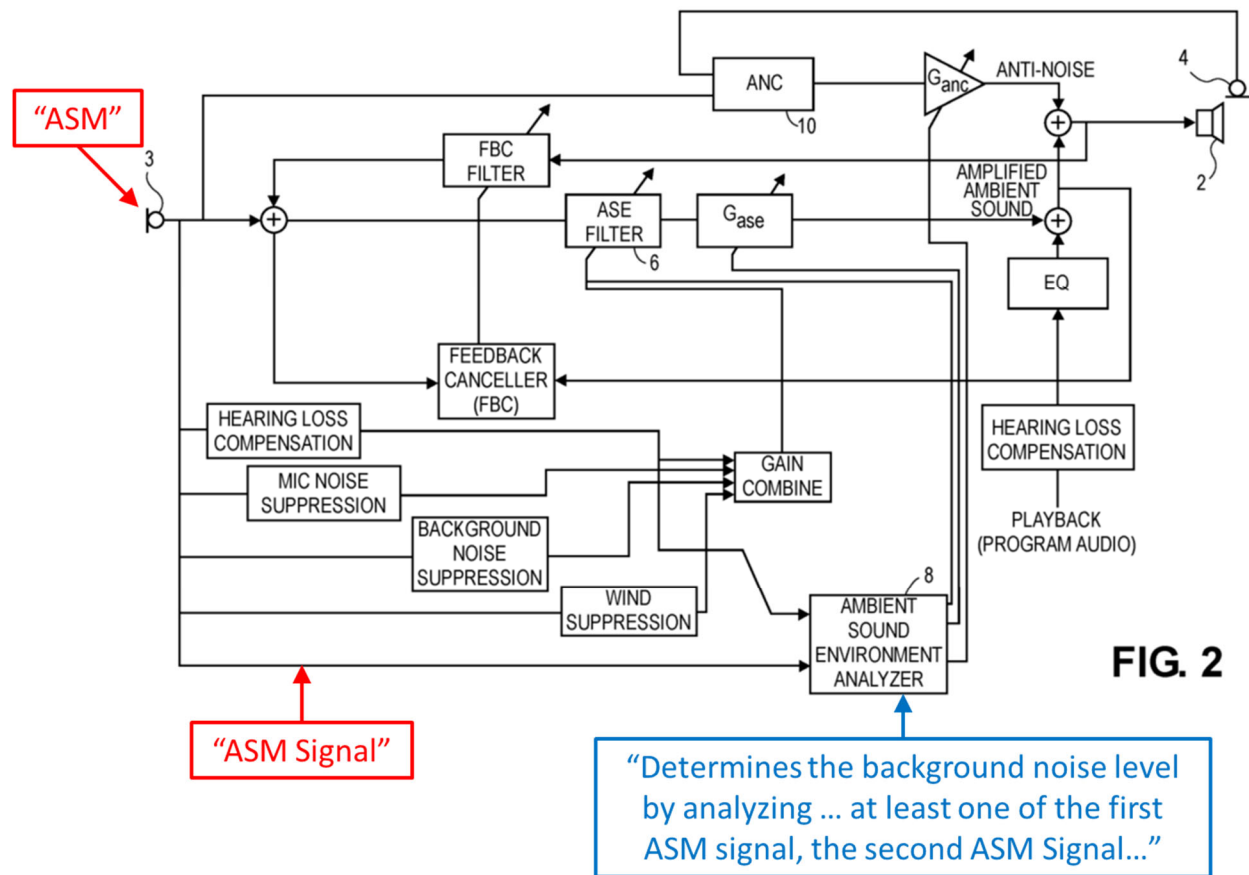
The *Usher-1-Woodruff-1/Woodruff-2* combination discloses/suggests this claim. *Kyriakakis*, ¶¶179-182. *Woodruff-1* discloses determining a “background noise level.” *Woodruff-1* discloses that “[w]hen the ambient sound enhancement function is active (the ambient sound is being reproduced, as described above), the

ambient sound environment analyzer 8 may detect that the ambient sound environment changes from quiet to loud.” *Woodruff-1*, ¶[0026]. *Woodruff-1* further discloses that “[a] noise profile analyzer, an automatic sound event detector, or a sound environment detector (any of which may be based on a machine learning model) may be used for this purpose, to distinguish between a loud environment and a quiet or active talker environment,” and that “[t]he loud environment may, for example, be one where the SPL is above a threshold, such that the audio system should perform an active earplug function to reduce the SPL at the user's ears to more comfortable levels.” *Id.*

POSITAs would have understood that this is a determination of the “background noise level,” as *Woodruff-1* describes using a “noise profile analyzer” to distinguish between loud and quiet environments. *Kyriakakis*, ¶180. Indeed, that *Woodruff-1* refers to a loud environment as compared to a “quiet environment or active talker environment” demonstrates that it is calculating a “background noise level.” In fact, in paragraph [0028], *Woodruff-1* discloses that its ambient sound environment analyzer 8 computes a sound pressure level.

Woodruff-1 further discloses that its ambient sound environment analyzer 8 uses the signal output by its ASM microphone to determine the background noise level. *Woodruff-1*, ¶[0028] (...the ambient sound environment analyzer 8 may simply compute SPL as derived from the signal from the external microphone 3...).

As discussed in Section IX.A.1.k, *Woodruff-1*'s microphone 3 corresponds to a claimed ASM and its output corresponds to a claimed "ASM signal." This is shown in annotated Fig. 2:



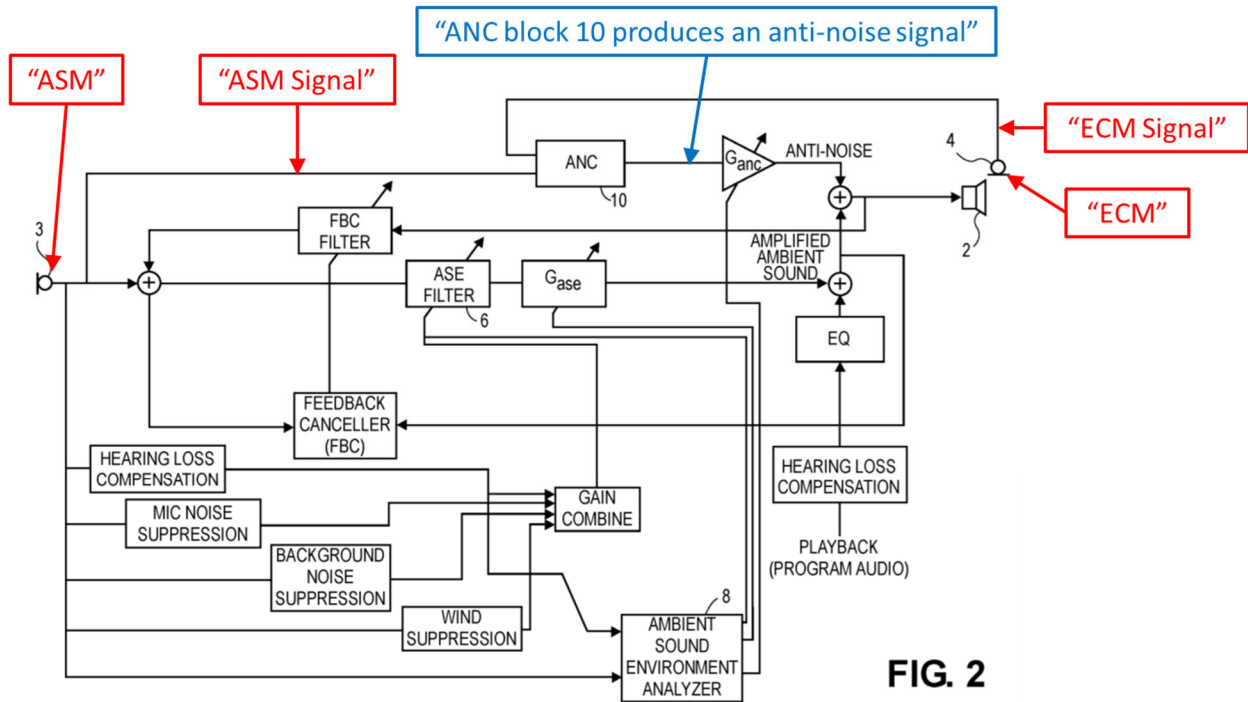
Kyriakakis, ¶181. Thus, the *Usher-1-Woodruff-1/Woodruff-2* combination renders claim 9 obvious.

9. Claim 11

The *Usher-1-Woodruff-1/Woodruff-2* combination discloses/suggests this claim. *Kyriakakis*, ¶¶183-185. Claim 11, which depends from claim 9, further requires generating an anti-noise signal using one of the ASM signals or the ECM

signal. *Woodruff-1* discloses that its “ANC block 10 produces an anti-noise signal while the ambient sound enhancement function is active.” *Woodruff-1*, ¶[0029]. *Woodruff-1* further discloses that its “ANC block 10 is shown in this example as being based on a hybrid feedforward and feedback algorithm (using input from both the external microphone 3 (also referred to as the reference microphone) and the internal microphone 4 (also referred to as the error microphone.)). *Id.* As discussed in Section IX.A.1.k, *Woodruff-1*’s microphone 3 corresponds to a claimed ASM and its output corresponds to a claimed “ASM signal.” POSITAs would have further understood that *Woodruff-1*’s microphone 4 corresponds to the claimed “ear canal microphone,” and the output of that microphone corresponds to the claimed “ECM signal.” *Kyriakakis*, ¶183.

Thus, this claim is disclosed by *Woodruff-1*, which has been seen in this annotated Fig. 2:

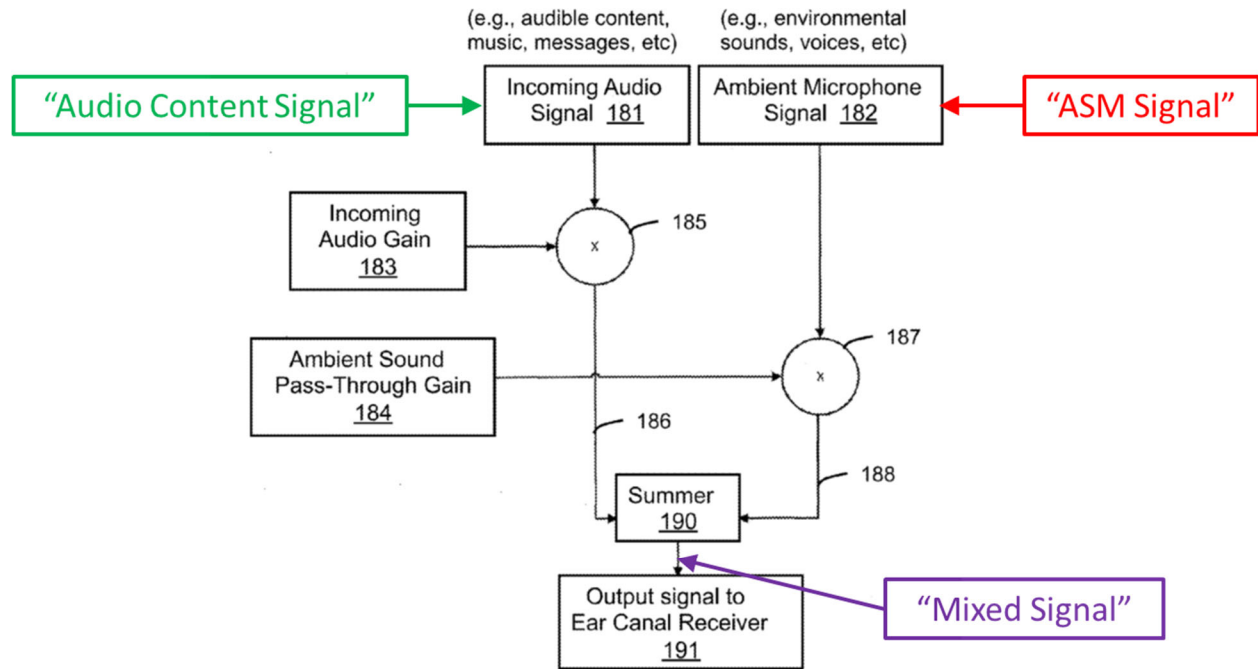


Kyriakakis, ¶184. Thus, the *Usher-1-Woodruff-1/Woodruff-2* combination renders claim 9 obvious.

10. Claim 13

The *Usher-1-Woodruff-1/Woodruff-2* combination discloses/suggests this claim. *Kyriakakis*, ¶¶186-189. *Usher-1* discloses mixing its incoming audio signal with signals from its ASMs 101/102. See *Usher-1*, ¶[0006] (“FIG. 1B illustrates a flowchart of a method for enhancing situation awareness of the earphone in FIG. 1A by way of mixing ambient sound signals and audible content in accordance with an exemplary embodiment.”), ¶[0022] (“Based on an analysis of the at least two ambient microphone (101/102) signals, at least one of the ambient microphone signals is mixed with the incoming audio signal 107 and directed to the ear canal

receiver (ECR) 105 (i.e., a loudspeaker in the sound isolating earphone that audibly delivers the audible signal to the user's eardrum).”). *See also* annotated Fig. 1B:



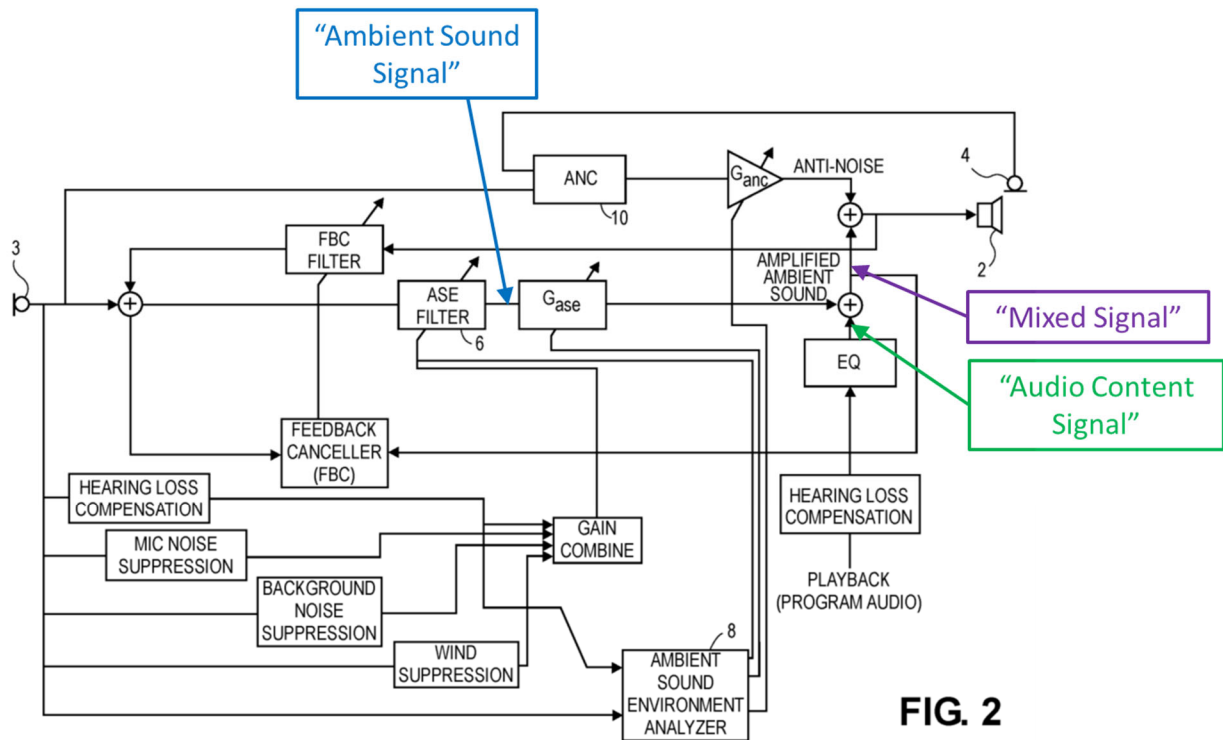
180
FIG. 1B

As discussed in Section IX.A.1.k, *Usher-1* does not explicitly disclose the claimed “ambient sound signal,” but *Woodruff-1* does. POSITAs would have recognized that *Woodruff-1*, like *Usher-1*, discloses an “audio content signal. *See Woodruff-1*, ¶[0025] (“The ambient sound content is then combined with a playback signal which contains program audio. The program audio may be music, the voice of a far end user during a phone call, or the voice of a virtual assistant program, which has not yet been converted to sound in the ambient environment.”). POSITAs would have

recognized that *Woodruff-1*'s "program audio" corresponds to the claimed "audio content signal." *Kyriakakis*, ¶187.

Woodruff-1 further discloses that its "program audio is mixed with its audio content signal. See *Woodruff-1*, ¶[0025] ("The ambient sound content is then combined with a playback signal which contains program audio."). See also *Id.*, ¶[0004]. POSITAs would have understood that *Woodruff-1*'s combined signal corresponds to the claimed "mixed signal." *Kyriakakis*, ¶188. This is also seen in annotated Fig. 2:¹⁷

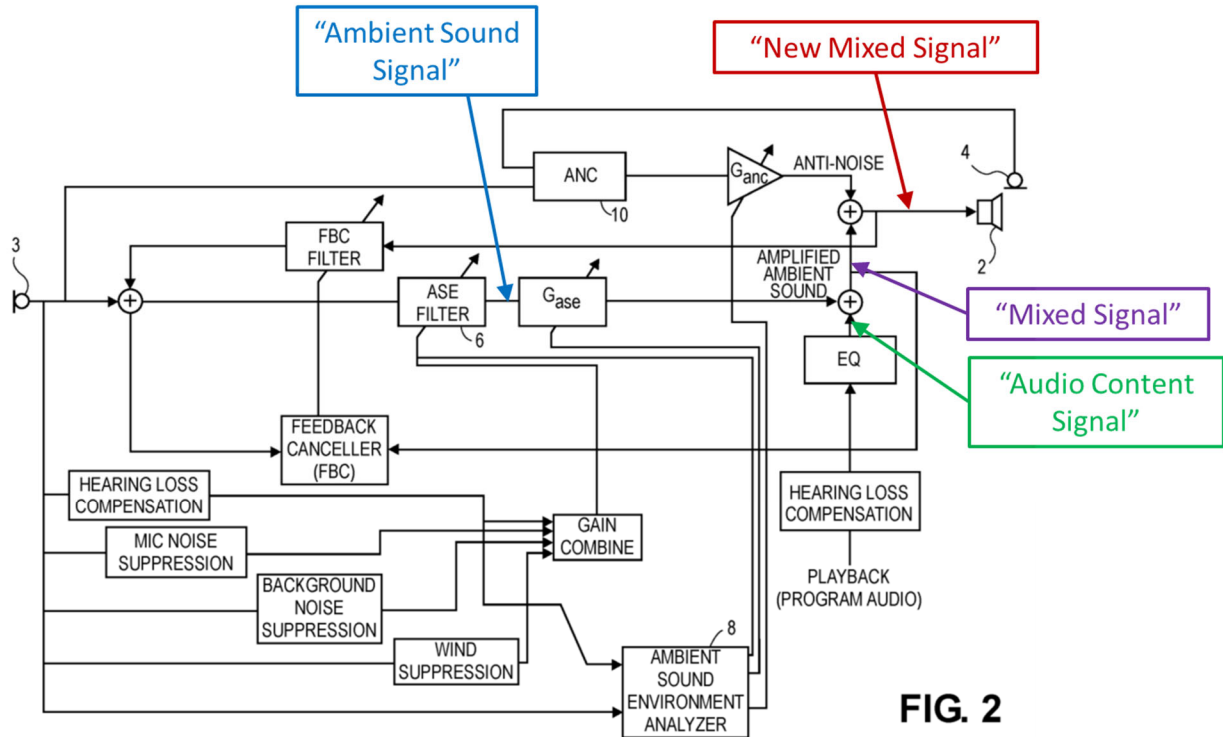
¹⁷ Note that an amplified version of *Woodruff-1*'s "ambient sound signal" is what is mixed with its "audio content signal." POSITAs would have understood that the amplified "ambient sound signal" is still the claimed "ambient sound signal" since it contains the same information. *Kyriakakis*, Footnote 11.



Thus, the *Usher-1-Woodruff-1/Woodruff-2* combination renders claim 13 obvious.

11. Claim 14

The *Usher-1-Woodruff-1/Woodruff-2* combination discloses/suggests this claim. *Kyriakakis*, ¶¶190-192. *Woodruff-1* discloses that the signal formed by combining its ambient sound signal with its program audio signal, (which as discussed in Section IX.A.10, corresponds to the claimed “mixed signal”) is further combined with an anti-noise signal. This is shown in annotated Fig. 2:



As can be seen, the “mixed signal” is combined with the anti-noise, and this signal is output to speaker 4. POSITAs would have understood that this combined signal corresponds to the claimed “new mixed signal.” *See Woodruff-1*, ¶[0004] (“When the ambient sound enhancement function is active, the input audio signal contains ambient or environmental sound pick up from one or more microphones in the device; in a “playback” situation, the input audio signal is a combination signal that also contains program audio, such as music, the voice of a far end user during a phone call, or the voice of a virtual assistant program.”); ¶[0029] (“The ANC block 10 produces an anti-noise signal while the ambient sound enhancement function is active: the anti-noise signal is combined with the ambient sound signal before they together drive the speaker 2 of the against the ear audio device.”). *See also*

Kyriakakis, ¶191. Thus, the *Usher-1-Woodruff-1/Woodruff-2* combination renders claim 14 obvious.

12. Claim 15

The *Usher-1-Woodruff-1/Woodruff-2* combination discloses/suggests this claim. *Kyriakakis*, ¶¶193-194. *Woodruff-1* discloses a situation where a user may be at a concert when due to a lull in the performance, a user starts talking. In this situation, *Woodruff-1* discloses that its device “...may reduce its anti-noise output (e.g., raise attenuation by G_{anc} , and perhaps increase its transparency gain G_{ase} that it applies to an ambient sound pickup channel” See *Woodruff-1*, ¶[0044]; ¶[0033] (“Another aspect of the disclosure here is a method performed by an audio system for speech-aware sound enhancement by an against the ear audio device, where the device can produce anti-noise for acoustic noise cancellation, ANC. In such a method, ambient sound is amplified and reproduced by the against the ear audio device in accordance with a hearing profile of the user of the against the ear audio device... While reproducing the ambient sound, the processor detects speech in the ambient sound (e.g., via voice activity detection, VAD, operating upon the external microphone signal). While reproducing the ambient sound, the against the ear device also produces anti-noise; it does that so long as no speech is detected in the ambient sound; whenever speech is detected, it reduces a level of the produced anti-noise. This enables the user to better perceive the speech in the reproduced ambient

sound.”). *Kyriakakis*, ¶193. Thus, *Woodruff-1* discloses the subject matter of claim 15, meaning that the *Usher-1-Woodruff-1/Woodruff-2* combination renders this claim obvious.

13. Claim 19

a. [19pre]

See Section IX.A.1.a.

b. [19.1]

See Section IX.A.1.b.

c. [19.2]

See Section IX.A.1.c.

d. [19.3]

See Section IX.A.1.d.

e. [19.4]

See Section IX.A.1.e.

f. [19.5]

See Section IX.A.1.f.

g. [19.6]

See Section IX.A.1.g.

h. [19.7]

See Sections IX.A.1.h-IX.A.1.j.

i. [19.8]

See Section IX.A.1.k.

j. [19.9]

See Section IX.A.1.l.

k. [19.10]

See Section IX.A.1.m.

l. [19.11]

See Section IX.A.1.n. Note that this element recites adjusting a “level of the ambient sound signal,” whereas element [1.13] is silent as to the “level.” POSITAs would have understood that *Woodruff-1* discloses adjusting the “level” of the “ambient sound signal” since it applies a gain using G_{ase} . *Kyriakakis*, ¶¶206-207. POSITAs would have known that applying a gain adjusts the level of a signal. *Id.* Thus, the *Usher-1-Woodruff-1/Woodruff-2* combination renders claim 19 obvious.

14. Claim 20

See Section IX.A.2. Note that claim 2 requires adjusting the “audio content signal” by reducing its volume if user speech is detected. Claim 19 only recites adjustment of “audio content,” and does not require a volume reduction. Regardless, POSITAs would have understood that the volume reduction *Usher-1* describes is an adjustment to “audio content,” just as claimed. *Kyriakakis*, ¶208.

15. Claim 21

a. [21pre]

Usher-1 discloses a method. *See e.g.*, Abstract. The same is true for *Woodruff-1* (e.g., ¶[0009]) and *Woodruff-1* (e.g., 2:49-56). Thus, the *Usher-1-Woodruff-1/Woodruff-2* discloses the preamble.

b. [21.1]

See Sections IX.A.1.b and IX.A.1.h.

c. [21.2]

See Sections IX.A.1.c and IX.A.1.i.

d. [21.3]

See Sections IX.A.1.d and IX.A.1.j.

e. [21.4]

See Section IX.A.1.k

f. [21.5]

See Section IX.A.1.l.

g. [21.6]

See Section IX.A.1.m. Note that element [21.6] requires detecting if the user is speaking by analyzing “the first and the second ASM signal,” whereas element [1.12] discussed in Section IX.A.1.m only requires analyzing one of them. Regardless, *Usher-1* discloses analyzing both. *Usher-1*, ¶[0030] “...processor 103

monitors sound from the ambient sound microphone(s) 101/102 and determines when user voice activity is detected.” *Id.*, at ¶[0030]. *Woodruff-1* also discloses this element. *See Woodruff-1*, ¶[0033] and ¶[0037] (“The user context encompasses any one or more the following. In one aspect, the determined user context discriminates between different types of ambient sound environments in which the wearer can find themselves, e.g., a concert, a public transport arena, a loud social gathering at a restaurant, in a coffee shop, in an office, and outdoors. ... The input data may include an ambient sound pickup channel (from one or more microphones in the against the ear audio device...”). *Kyriakakis*, ¶215. Thus, the *Usher-1-Woodruff-1/Woodruff-2* combination discloses this element.

h. [21.7]

The *Usher-1-Woodruff-1/Woodruff-2* combination discloses this element. This element recites that the “mixed signal” is a “combination of the ambient sound signal, the audio content signal, and an anti-noise signal.

As discussed in Section IX.A.1.n, when *Usher-1* detects that a user is speaking, “the gain of the incoming audio signal is decreased at step 253...” *Usher-1*, ¶[0030]. As discussed in Section IX.A.1.n, *Usher-1*’s “incoming audio signal corresponds to the claimed “audio content signal.”

As also discussed in Section IX.A.1.n, when *Usher-1* detects that a user is speaking, “...the ambient sound pass-through gain is increased at step 254.” *Usher-*

1, ¶[0030]. See also *Id.*, ¶[0040]-[0041]. As discussed in Section IX.A.1.k, when *Usher-1* is combined with *Woodruff-1/Woodruff-2*, the ASM signal produced by ASM microphone 101/102 is combined with the output of *Woodruff-1/Woodruff-2*'s FBC Filter 10 to produce the claimed “first signal,” which in turn is filtered by ASE Filter 6 to produce the claimed “ambient sound signal.” *Kyriakakis*, ¶219. *Woodruff-1* is similar to *Usher-1*, as it also discloses that the signal output by its ASE Filter 6, corresponding to the claimed “ambient sound signal” can be adjusted when the user speaks. *Woodruff-1*, ¶¶[0042]-[0044]; see also *id.*, ¶¶[0033], [0049].

Finally, as discussed in Section IX.A.12, *Woodruff-1* discloses that when a user starts talking, its device “...may reduce its anti-noise output (e.g., raise attenuation by G_{anc}).” See *Woodruff-1*, ¶[0044].

Note that in the *Usher-1-Woodruff-1/Woodruff-2* combination, these three signals are adjusted prior to mixing. However, POSITAs would have understood that the act of adjusting these three signals would result in the claimed step of “adjusting a mixed signal sent to the speaker” because the components combined into the “mixed signal” would be adjusted when a user starts speaking, and then mixed. The resulting “mixed signal” would not be the same, i.e., would be “adjusted,” when compared to the mixed signal sent the speaker prior to the user speaking. *Kyriakakis*, ¶221. Indeed, there is no claim language requiring that any action, e.g., application of gain, be applied to the combined signal. All that is

required is that the mixed signal be adjusted.

That the mixed signal can be adjusted by adjusting its input signals is confirmed by claim 22, which depends on claim 21, and which recites that mixed signal is adjusted “by increasing the volume of the audio content signal.” *Id.* Any construction of limitation [21.7] requiring that the mixed signal itself be acted on would be inconsistent with claim 22, which is not how claims are normally construed. *Multilayer Stretch Cling Film Holdings, Inc. v. Berry Plastics Corp.*, 831 F.3d 1350, 1362 (Fed. Cir. 2016) (“[a] dependent claim that contradicts, rather than narrows, the claim from which it depends is invalid.”).

The *Usher-1-Woodruff-1/Woodruff-2* combination renders this claim obvious. *Kyriakakis*, ¶223.

16. Claim 22

See Section IX.A.2.

17. Claim 23

See Section IX.A.4.

18. Claim 25

See Section IX.A.6

19. Claim 26

See Section IX.A.7.

20. Claim 27

See Section IX.A.8.

21. Claim 29

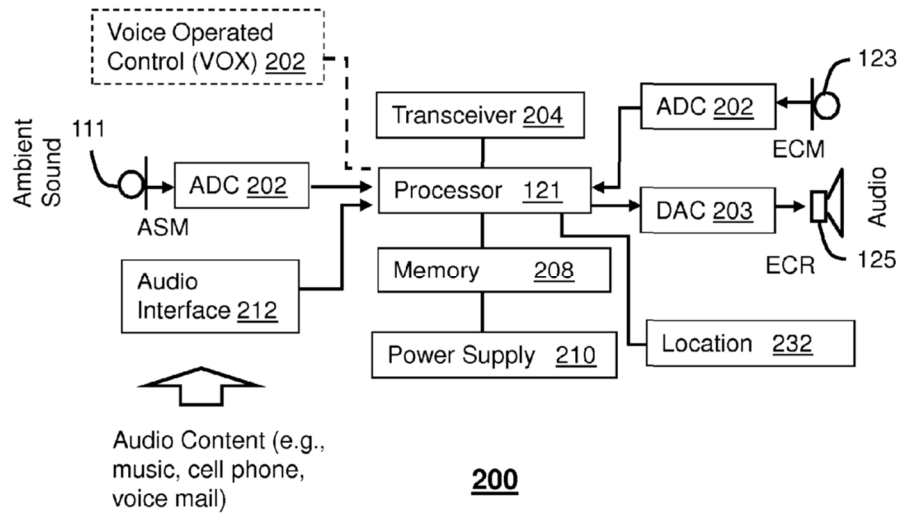
See Section IX.A.9.

B. Ground 2: *Usher-1*, *Woodruff-1*, *Woodruff-2* and *Usher-2* Render Obvious Claims 5, 24

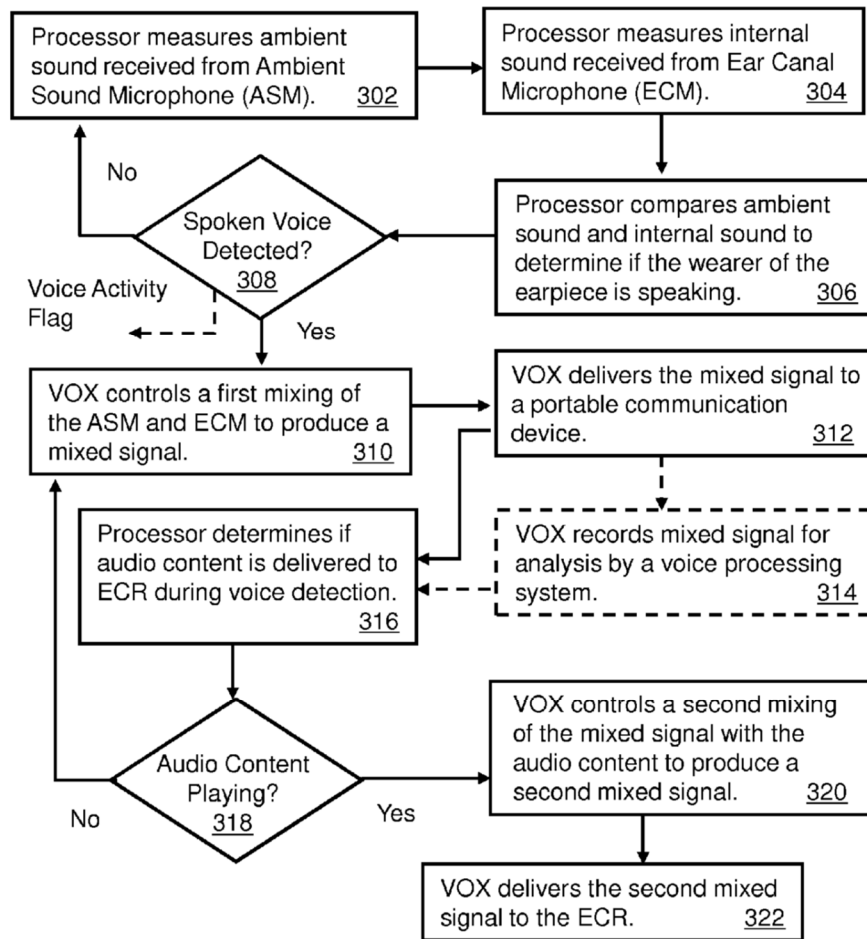
1. Claim 5

The *Usher-1-Woodruff-1/Woodruff-2-Usher-2* combination discloses or suggests this claim. *Kyriakakis*, ¶¶230-235. Claim 5 requires analyzing one or more ASM and/or ECM signals to detect a voice command, and requires that this analysis take place “if the user speaking is detected.” *Id.*, ¶230. As discussed in Section IX.A.1.m, *Usher-1* discloses analyzing ASM signals to determine whether the user is speaking. *Usher-1*, ¶[0030]. *Usher-1* further discloses that its system can implement “smart” functions for “...identifying voice commands from a user...” *Id.*, ¶[0050]. *Usher-1*, however, does not explicitly disclose a device where the identification of user’s voice command is performed by the earphone itself or how such voice identification is performed.

Usher-2 discloses such device. *Usher-2* discloses an earphone similar to the one described in the ’965*Pat*, as it has an ASM 111, and ECM 123, a processor 121 and a speaker 125. *Usher-2*, ¶¶[0030]-[0032]. *See also* Fig. 2:



Usher-2's process for detecting voice commands is illustrated in Fig. 3:



300
FIG. 3

Usher-2 discloses that at step 306, its processor “determines if the wearer...of the earpiece 100 is speaking.” *Usher-2*, ¶[0046]. If user speech is detected (step 308), then *Usher-2*’s voice operated control (“VOX”) 201 mixes the signals from its ASM and ECM to produce a mixed signal (step 310). *Id.*, ¶[0047]. *Usher-2* discloses that that VOX 201 can “record the mixed signal for further analysis by a voice processing system,” which, at step 314, “allows the processor 121, or voice processing system receiving the mixed signal to analyze the mixed signal for information, such as voice

commands.” *Id.*, ¶[[0055].

Claim 5 requires that detecting voice commands by “analyzing at least one of the first ASM signal, the second ASM signal, the ECM signal, or a combination thereof.” *Usher-2* discloses detecting voice commands using a mixed signal generated by combining its ASM and ECM signals. Note also that that *Usher-2* discloses that its processor operates by executing instructions stored in memory. *Id.*, ¶[[0037]. Thus, *Usher-2* discloses claim 5

**a. POSITAs Would Have Been Motivated To Combine
Usher-2 with *Usher-1-Woodruff-1/Woodruff-2***

As discussed, *Usher-1* suggests that its system can implement “smart” functions for “...identifying voice commands from a user...” *Id.*, ¶[[0050]. *Usher-1*, however, does not describe how to implement voice command identification. *Usher-1*’s suggestion, however, would have motivated POSITAs to look to other teachings for how to implement voice command identification, which would have led such POSITAs to *Usher-2*’s solution. *Kyriakakis*, ¶235. Indeed, *Usher-1* and *Usher-2* have two common inventors, and *Usher-1* incorporates *Usher-2* by reference. See Section V.D. POSITAs would have recognized that *Usher-2*’s voice command identification technique provided acceptable performance while implementing into *Usher-1-Woodruff-1/Woodruff-2*, and adding *Usher-2* to *Usher-1* would have been well within the skill of an ordinary artisan. *Kyriakakis*, ¶235.

2. Claim 24

See Section IX.B.1.¹⁸

**C. Ground 3: *Usher-1*, *Woodruff-1*, *Woodruff-2* and *Nicholson*
Render Obvious Claims 10, 12, 16-18, 28, 30**

1. Claim 10

The *Usher-1-Woodruff-1/Woodruff-2-Nicholson* combination discloses or suggests this claim. *Kyriakakis*, ¶¶237-251. The only difference between claim 10 and claim 9 is that claim 10 requires determining the background noise level by analyzing modified versions of the first ASM, second ASM, or ECM signals (or a combination thereof). *Id.*, ¶237. *Woodruff-1* does not explicitly disclose use of “modified” signals to determine its background noise signal. However, *Nicholson*, which has the same assignee as *Woodruff-1*, contains such teachings.

Nicholson discloses an ANC technique (*Nicholson*, ¶[0004]) in an earphone similar to that of the ’965*Pat* and the prior art, as it also comprises an ECM 8 (which

¹⁸ On 10/23/2023, PO requested a Certificate of Correction that makes claim 24 dependent on claim 21, not non-existent claim 31. The claim is obvious for the reasons described in Section IX.B.1 even if dependent on claim 21. *Kyriakakis*, ¶236.

it calls an error microphone”), and loudspeaker 6:¹⁹

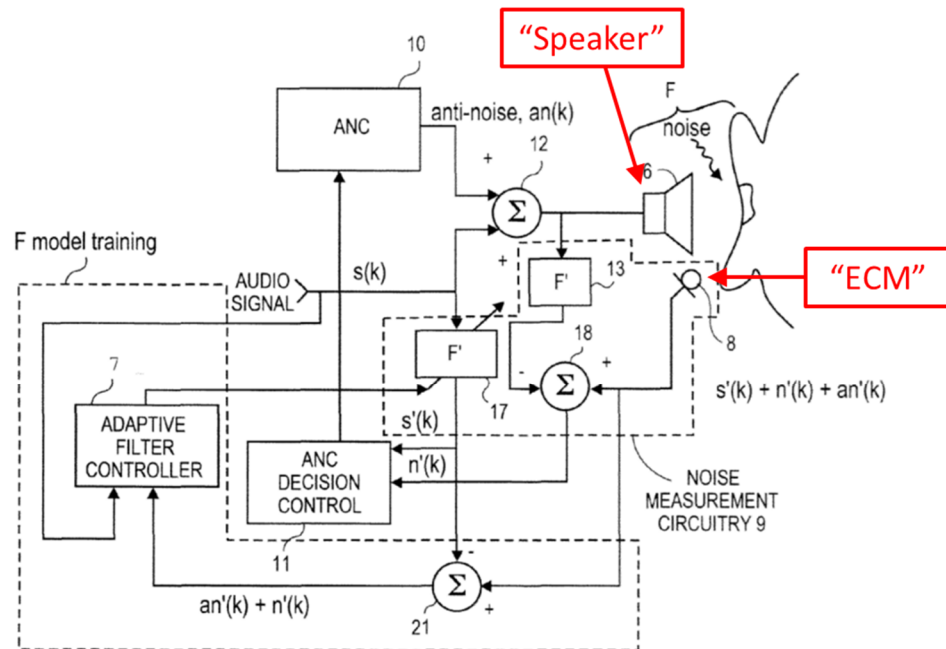


FIG. 2

Nicholson, Fig. 2. *Kyriakakis*, ¶238. *Nicholson's* earphone has ANC Decision Control 11 that “determines whether to activate or deactivate the ANC block 10, based on computed or estimated values for signal, $S'(k)$, and noise, $n'(k)$.” *Id.*, ¶[0026]. As will be discussed in Section IX.C.1.a, use of a decision control 11 to determine when to activate ANC block 10 will improve the functionality of the *Usher-1-Woodruff-1/Woodruff-2* combination.

Nicholson further discloses that “signal and noise estimates are computed by

¹⁹ *Nicholson* also describes an embodiment having both an ECM and an ASM (reference microphone 9). See *Nicholson*, Fig. 5.

noise measurement circuitry 9, which includes an error microphone 8 that is located and oriented in such a manner as to pickup both (a) sound emitted from the earpiece speaker 6 and (b) the ambient acoustic noise that has leaked into the cavity or region between the handset housing or shell (not shown) that is in front of the earpiece speaker 6 and the user's ear.” *Id.*, ¶[0027]. *Kyriakakis*, ¶240.

Nicholson further discloses that its ECM 8 outputs a signal. *Id.* (“A signal picked up by the microphone...”). POSITAs would have recognized that this signal corresponds to the claimed “ECM signal.” *Kyriakakis*, ¶241. *Nicholson* discloses that this ECM signal “is fed to a differencing unit 18 whose other input receives a signal from the output of the first filter 13.” *Nicholson*, ¶[0027]. *Nicholson* states that differencing unit 18 “provide[s] an estimate of the ambient acoustic noise $n'(k)$...” *Id.* Because differencing unit 18 modifies the signal output by ECM 8 by subtracting the signal F' , POSITAs would have understood that differencing unit 18’s output $n'(k)$ corresponds to the claimed “modified ECM signal.” *Kyriakakis*, ¶241. This is seen in annotated Fig. 2:

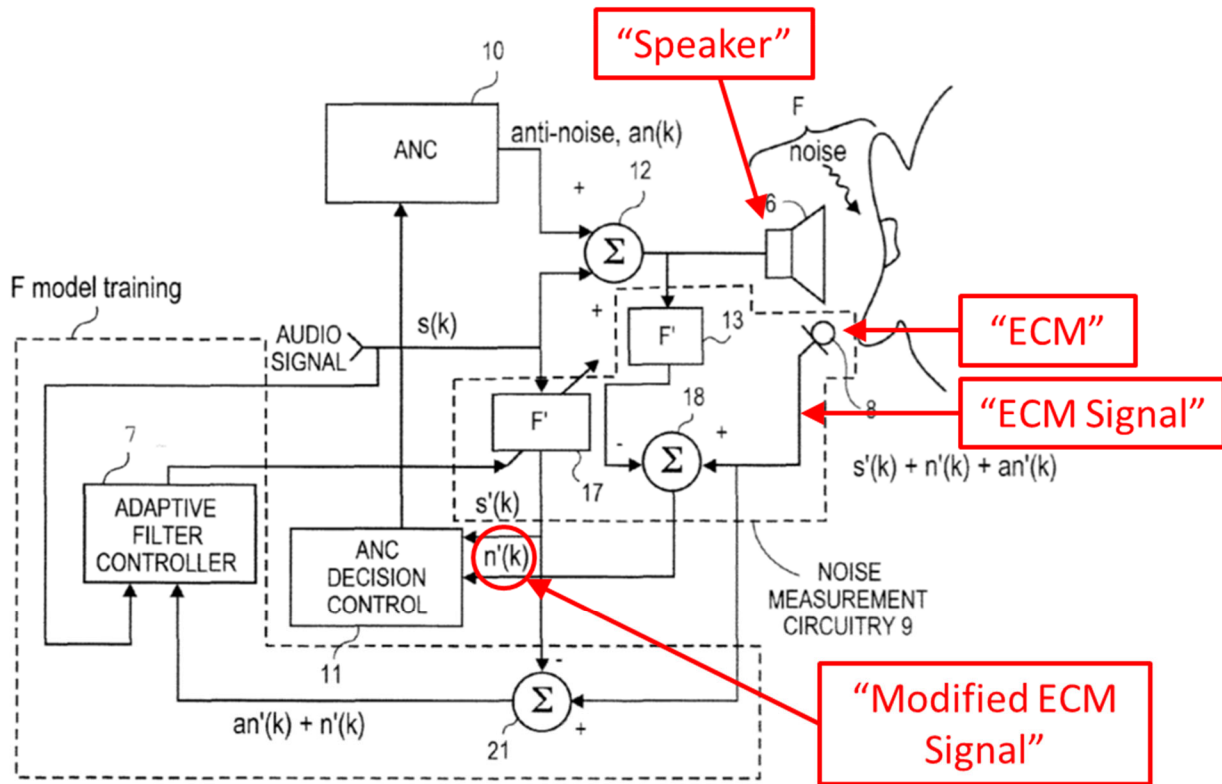


FIG. 2

Kyriakakis, ¶241.

Nicholson's modified ECM signal $n'(k)$ (along with “desired audio signal $s'(k)$ ”) is input to ANC Decision Control 11. *Nicholson*, ¶[0028] (“The estimated signals $s'(k)$ and $n'(k)$ are input to the ANC decision control circuitry 11, which can then determine an estimate of how much sound emitted from the earpiece speaker 6 has been corrupted by the ambient acoustic noise (e.g., SNR).”). The algorithm for *Nicholson's* decision control circuitry 11 is shown in its Fig. 3, which shows that signals $n'(k)$ and $s'(k)$ “pass through a smoothing conditioner, which in this case includes a subjective loudness weighting block 12 and an averaging block 14.” *Id.*

at [0032]. *Nicholson's* loudness weighting block 12 “may be a typical filtering operation used when measuring noise in audio systems (e.g., A-weighting, ITU-R 468).” *Id.* Next, *Nicholson's* “averaging block 14 may implement a typical root mean square or other suitable signal averaging algorithm, e.g. ITU-T G.160...” *Id.* *Kyriakakis*, ¶242.

Note that signal $s'(k)$, the desired audio signal, goes through the same process (the bottom branch of the ANC Decision Control 11 of *Nicholson's* Fig. 3). The output of averaging block 14 for the noise and the signal are input to threshold decision block 15, which “compute[s] the signal to noise ratio by essentially comparing the smoothed noise estimate $n''(k)$ to the smoothed signal estimate $s''(k)$. *Kyriakakis*, ¶243.

POSITAs would have understood that the signal output by averaging block 14, $n''(k)$, corresponds to the claimed “background noise level,” since it is a “smoothed noise estimate,” which POSITAs would have understood represents the “background noise level. This is seen in annotated Fig. 3:

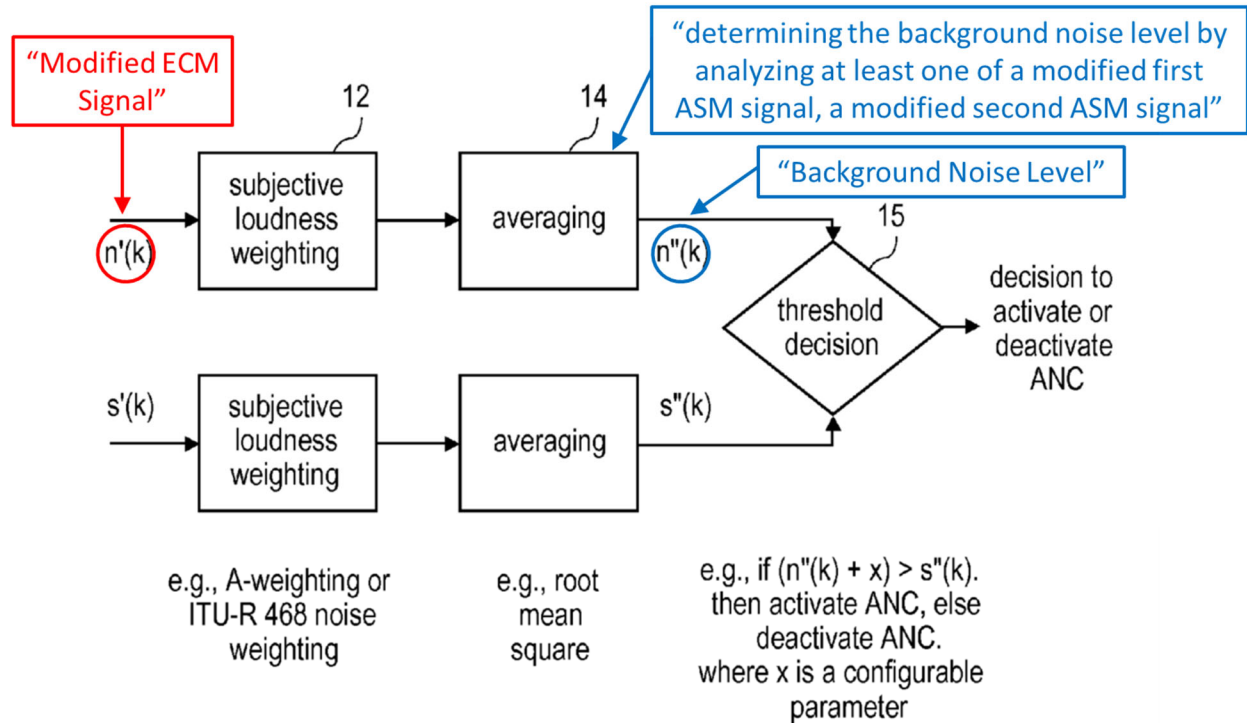


FIG. 3

Kyriakakis, ¶244. Calculating the signal-to-noise ratio “essentially determines whether the sound being emitted from the earpiece speaker 6 has been sufficiently corrupted by the ambient acoustic noise...” *Nicholson*, ¶[0033]. In *Nicholson’s* earphone, “[i]f the SNR is below a configurable parameter or threshold, then the decision is made to not deactivate the ANC circuitry, or to activate it.” *Id.* The reason for this is that when the signal-to-noise ratio is low, “it is expected that ANC is likely to achieve some substantial reduction in the unwanted sound that the user may be hearing.” *Id.* In contrast, when the signal-to-noise ratio is high, “then this suggests that the ambient acoustic environment may be sufficiently quiet such that ANC is likely to provide no benefit to the user and hence should be deactivated or

disabled, or not activated or enabled.” *Id.* Keeping the ANC circuitry off, or deactivating it if on, “save power and avoid unwanted audio artifacts.” *Id.* Thus, POSITAs would have recognized that *Nicholson*’s will improve earphones having ANC, meaning that *Nicholson* discloses claim 10 and renders it obvious. *Kyriakakis*, ¶245.

**a. POSITAs Would Have Been Motivated To Combine
Nicholson with *Usher-1-Woodruff-1/Woodruff-2***

The *Usher-1-Woodruff-1/Woodruff-2* combination discloses an earbud having acoustic noise cancellation (ANC). *See e.g., Woodruff-1*, Abstract. *See also* Sections IX.A.1.k and IX.A.8. POSITAs would have recognized that *Nicholson* sets out to improve the ANC function of earbuds, functionality also described in *Woodruff-1*. For example, *Nicholson* discloses that its method of determining when the ANC is activated helps alleviate “hiss” and reduce other artifacts of the operation of ANC circuitry. *Nicholson*, ¶[0007]. *See also id.*, *Nicholson* further discloses that this method reduces power consumption. *Id.* ¶[0025] (“The various embodiments of the invention make decisions on activation and deactivation of ANC in a way that helps reduce the presence of such audible artifacts and conserves power, when it has been determined that the ANC would not be of substantial benefit to the user.”). *Kyriakakis*, ¶246.

As discussed in Section IX.C.1, determining the signal-to-noise ratio of the

sound played in the ear canal, as described in *Nicholson*, will allow an earphone to both save power and reduce playback of ANC artifacts in situations where ambient noise is low. *See also Kyriakakis*, ¶247.

Moreover, *Woodruff-1* discloses that its ANC block 10 uses a hybrid ANC approach, using both feedforward and feedback approaches. *Woodruff-1*, ¶[0029]. *Nicholson's* Fig. 2 embodiment uses a feedback approach, as it uses its ECM microphone 8. *Nicholson*, ¶[0029]. *See also Kyriakakis*, ¶248. Thus, POSITAs would have recognized that *Nicholson's* ANC switching could have been implemented into *Usher-1-Woodruff-1/Woodruff-2* without extensive redesign and with the expectation that the combination would work for its intended purpose. *Id.*

POSITAs would have understood that any ANC performance improvements would provide commercial and competitive advantages and would be motivated to seek out teachings that would improve their existing technology. *Kyriakakis*, ¶249. POSITAs searching for teachings that would improve *Usher-1-Woodruff-1/Woodruff-2's* ANC would have been led to teachings like those in *Nicholson* because their systems are architecturally similar, as discussed.

Due to the structural and conceptual similarity, POSITAs would have expected that incorporating *Nicholson's* teachings into *Usher-1-Woodruff-1/Woodruff-2* would have worked and would have improved the performance of the earphone. *Kyriakakis*, ¶250. Thus, claim 10 is rendered obvious by the *Usher-1-*

Woodruff-1/Woodruff-2-Nicholson combination.

2. Claim 12

The *Usher-1-Woodruff-1/Woodruff-2-Nicholson* combination discloses or suggests this claim. *Kyriakakis*, ¶¶252-256. Claim 12, which depends from claim 10, requires generating an anti-noise signal “using at least one of the modified first ASM signal, the modified second ASM signal, the modified ECM signal or a combination thereof.” As discussed in Section IX.C.1, *Nicholson* discloses that differencing unit 18 outputs a signal $n'(k)$ corresponding to the claimed “modified ECM signal.” As further discussed there, *Nicholson* discloses that its ANC circuitry 10 “generates an anti-noise signal, $an(k)$...” *Nicholson*, ¶[0026]. *Nicholson*’s Fig. 6 “shows a block diagram of feedback ANC.” *Id.*, ¶[0038].²⁰ *Nicholson* teaches that noise measurement circuitry 9 having ECM 8 outputs a signal, to differencing unit 18, which as discussed in Section IX.C.1, modifies the signal output by ECM 8 by subtracting the signal F' therefrom, thus resulting noise estimate, $n'(k)$. As also discussed in Section IX.C.1, $n'(k)$ corresponds to the claimed “modified ECM signal,” which is input to ANC Decision Control 11. *Nicholson*, ¶[0028]. As also discussed there, *Nicholson* uses the modified ECM signal $n'(k)$ to determine whether

²⁰ *Nicholson* states that its Fig. 2 embodiment can be used for both feedback and feed forward ANC. *Nicholson*, ¶[0026].

its ANC block 10 should be enabled/disabled. *Id.*, ¶[0033]. *Nicholson* discloses that its ANC block 10 “generates an anti-noise signal, $an(k)$...” *Id.*, ¶[0026]. Because ANC Block 10’s generation of anti-noise is enabled by using *Nicholson*’s modified ECM signal $n'(k)$, POSITAs would have recognized that *Nicholson* “generat[es] an anti-noise signal using ... the modified ECM signal,” just as claimed. See *Nicholson*’s Fig. 2 (annotated):

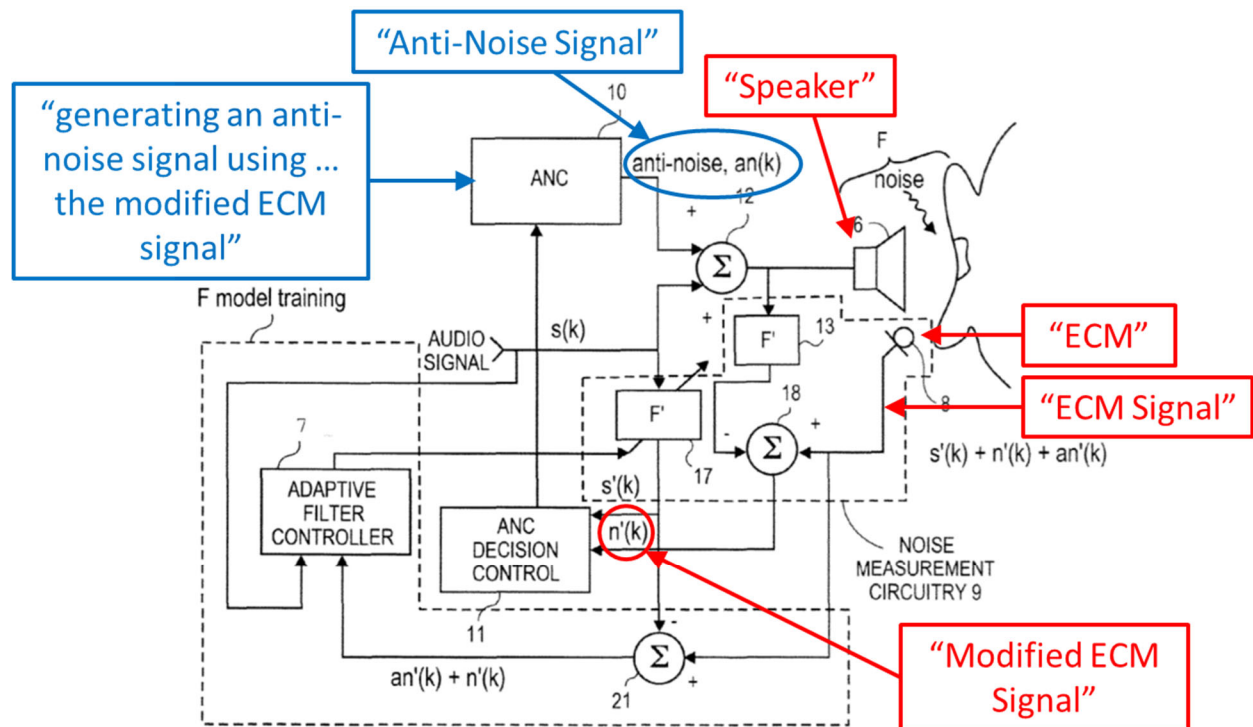


FIG. 2

POSITAs would have recognized that *Nicholson*’s ANC decision control and ANC blocks could easily be added to Woodruff-1, and would have been motivated to do so, as discussed in Section IX.C.1.a. See also *Kyriakakis*, ¶¶252-253.

Alternatively, *Nicholson* discloses that “the anti-noise signal input to the

mixer 12 is generated by an anti-noise digital filter 22 whose input is coupled to receive the noise estimate, $n'(k)$. *Nicholson*, ¶[0038]. Thus, POSITAs would have recognized that *Nicholson* discloses generating an anti-noise signal using a “modified ECM signal.” This can be seen in *Nicholson*’s Fig. 6 (annotated):

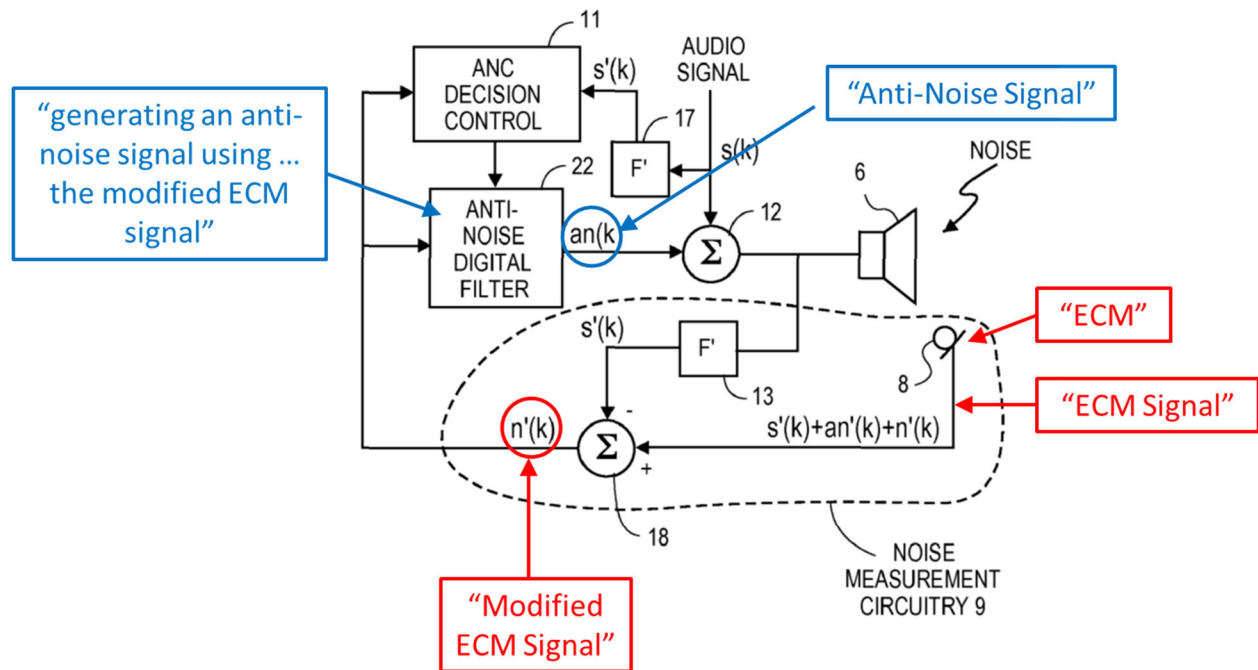


FIG. 6

Kyriakakis, ¶254.

As discussed, in Section IX.C.1 (and IX.C.1.a), POSITAs would have recognized that *Nicholson*’s feedback ANC techniques could be implemented into *Woodruff-1*’s hybrid ANC approach and would have been motivated to do so. POSITAs would have recognized that *Nicholson*’s ANC block 10, including its ANC Decision Control 11, could be implemented *Woodruff-1*’s ANC block 10 such that

the *Woodruff-1-Nicholson* combined ANC block used the modified ECM signal generated by *Nicholson*. This would allow the *Usher-1-Woodruff-1/Woodruff-2* device to reduce its power consumption prevent users from hearing ANC artifacts like hiss. *Kyriakakis*, ¶255. Thus, *Nicholson* discloses claim 12, meaning the *Usher-1-Woodruff-1/Woodruff-2-Nicholson* combination renders claim 12 obvious.

3. Claim 16

See Section IX.A.10.

4. Claim 17

See Section IX.A.11.

5. Claim 18

See Section IX.A.12.

6. Claim 28

See Section IX.C.1.

7. Claim 30

See Section IX.C.2.

X. SECONDARY CONSIDERATIONS

There is no evidence in the '965*pat*'s prosecution history or elsewhere supporting any secondary considerations arguments, or evidence of nexus of such alleged evidence to the challenged claims. *See* generally '965*FH*.

XI. THE DISCRETIONARY FACTORS FAVOR INSTITUTING TRIAL

The USPTO's June 2022 updated discretionary denial guidance expressly noted that the PTAB will not discretionarily deny institution when, as here, Petitioners present a Sotera stipulation. (*EX1015*).

XII. MANDATORY NOTICES UNDER 37 C.F.R. §42.8

A. Real Parties-in-Interest

Petitioners identify themselves as the real parties-in-interest.

1. Related Matters

To the best of Petitioners' knowledge, the '965pat has only been involved in the following district court litigation:

- *Staton Techiya, LLC v. Samsung Elecs. Co., Ltd. et al.*, 2:23-cv-00319-JRG-RSP (E.D. Tex.), filed July 10, 2023.

The following chart identifies all patents/application that are in the same family as the '965pat:

U.S. Patent App. No. 18/397,683, filed 12/27/2023
U.S. Patent App. No. 18/112,629, filed 02/22/2023
U.S. Patent App. No. 16/662,505, Now U.S. Patent No. 11,277,682
U.S. Patent App. No. 16/179,171, now U.S. Patent No. 10,506,329
U.S. Patent App. No. 14/109,987, now U.S. Patent No. 10,123,110

U.S. Patent App. No. 12/044,727, now U.S. Patent No. 8,625,812
U.S. Provisional App. 60/893,617

B. Lead and Backup Counsel

Lead Counsel	Back-Up Counsel
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C. Service Information

Please address all correspondence to lead and back-up counsel at the addresses shown above. Petitioners consent to electronic service by email at the following addresses:

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D. Power of Attorney

Powers of attorney are filed herewith according to 37 C.F.R. §42.10(b).

XIII. FEES

Petitioners concurrently electronically submits the required fees for this Petition. The Board is authorized to charge Arnold & Porter Kaye Scholer LLP's deposit account, No. 50-2387, for any fee deficiency.

Date: June 18, 2024

Respectfully submitted,

/Jeffrey A. Miller/

Jeffrey A. Miller (Reg. No. 35,287)
Counsel for Petitioners

CERTIFICATE OF COMPLIANCE

The undersigned hereby certifies that the foregoing Petition for *Inter Partes* Review contains 13,983 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.

/Jeffrey A. Miller/

Jeffrey A. Miller (Reg. No. 35,287)
Counsel for Petitioners

CERTIFICATE OF SERVICE

I certify that on June 18, 2024, I caused a true and correct copy of the foregoing Petition for *Inter Partes* Review of U.S. Patent No. 11,750,965 and supporting exhibits to be served via overnight delivery on the Patent Owner at the following correspondence address of record as listed on Patent Center:

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A courtesy copy was also sent via electronic mail to Patent Owner's litigation counsel listed below:

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