

Before The Patent Trial And Appeal Board
Oral Hearing
October 9, 2024

Jawbone's Presentation

IPR2023-01084: U.S. 8,321,213

IPR2023-01128: U.S. 8,326,611

All cites are to IPR2023-01084 and the '213 Patent



DEMONSTRATIVE EXHIBIT – NOT EVIDENCE

Claim Construction

“filter that describes a relationship for speech between the first physical microphone and the second physical microphone”

Jawbone’s Construction	Petitioner’s Construction
“A filter that alters a signal based on the non-temporal components of relative responses to speech of the first and second physical microphones”	No construction necessary.

Paper 29 at 10.

Claim Construction

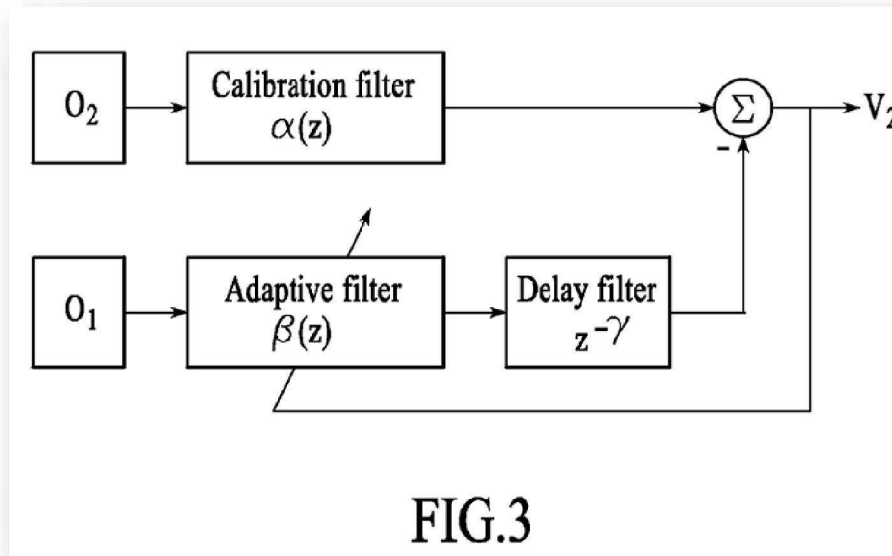
The Inventive Concept

The NAVSAD and PSAD systems include an array algorithm for speech detection **that uses the difference in frequency content between two microphones to calculate a *relationship between the signals of the two microphones***. This is in contrast to conventional arrays that attempt to use the time/phase difference of each microphone to remove the noise outside of an “area of sensitivity”. *The methods described herein provide a significant advantage, as they do not require a specific orientation of the array with respect to the signal.*

Paper 29 at 15 (quoting Ex. 1001, 11:20-29 (emphasis added))

Claim Construction

Three Filters Are Disclosed



Ex. 1001, Fig. 3
Paper 29 at 16

Claim Construction

Three Filters Are Disclosed

A better implementation is to use directional microphones where the second microphone has minimal speech response. As described herein, such microphones can be constructed using omnidirectional microphones O_1 and O_2 :

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

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

where $\alpha(z)$ is a calibration filter used to compensate O_2 's response so that it is the same as O_1 , $\beta(z)$ is a filter that describes the relationship between O_1 and calibrated O_2 for speech, and γ is a fixed delay that depends on the size of the array. There is no loss of generality in defining $\alpha(z)$ as above, as either microphone may be compensated to match the other.

Ex. 1001, 5:20-24
Paper 29 at 15-16

Claim Construction

Three Filters Are Claimed

phone, the processing component forming a filter that describes a relationship for speech between the first physical microphone and the second physical microphone, the processing component forming a second virtual microphone by applying the filter to the first signal to generate a first intermediate signal, and summing the

Ex. 1001, 38:1-6; Paper 29 at 13

15. The device of claim **14**, comprising applying a calibration to at least one of the first signal and the second signal.

16. The device of claim **15**, wherein the calibration compensates a second response of the second physical microphone so that the second response is equivalent to a first response of the first physical microphone.

Ex. 1001, 38:12-17; Paper 29 at 20

17. The device of claim **15**, comprising applying a delay to the first intermediate signal.

18. The device of claim **17**, wherein the delay is proportional to a time difference between arrival of the speech at the second physical microphone and arrival of the speech at the first physical microphone.

Ex. 1001, 38:18-23; Paper 29 at 20

Prior Art

Avendano

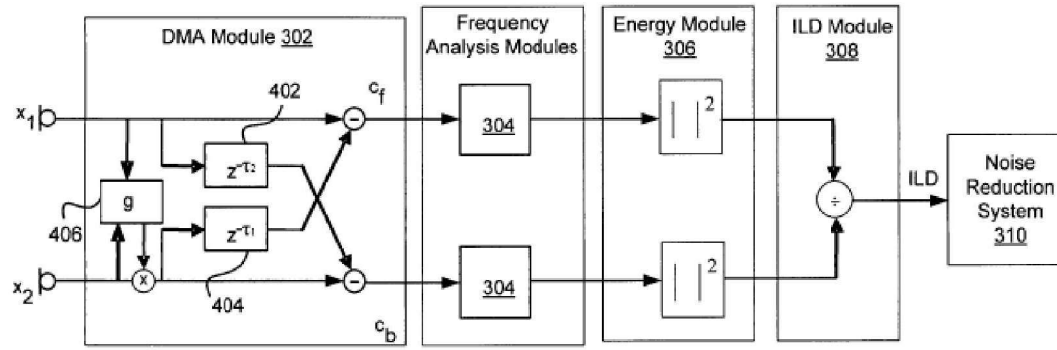
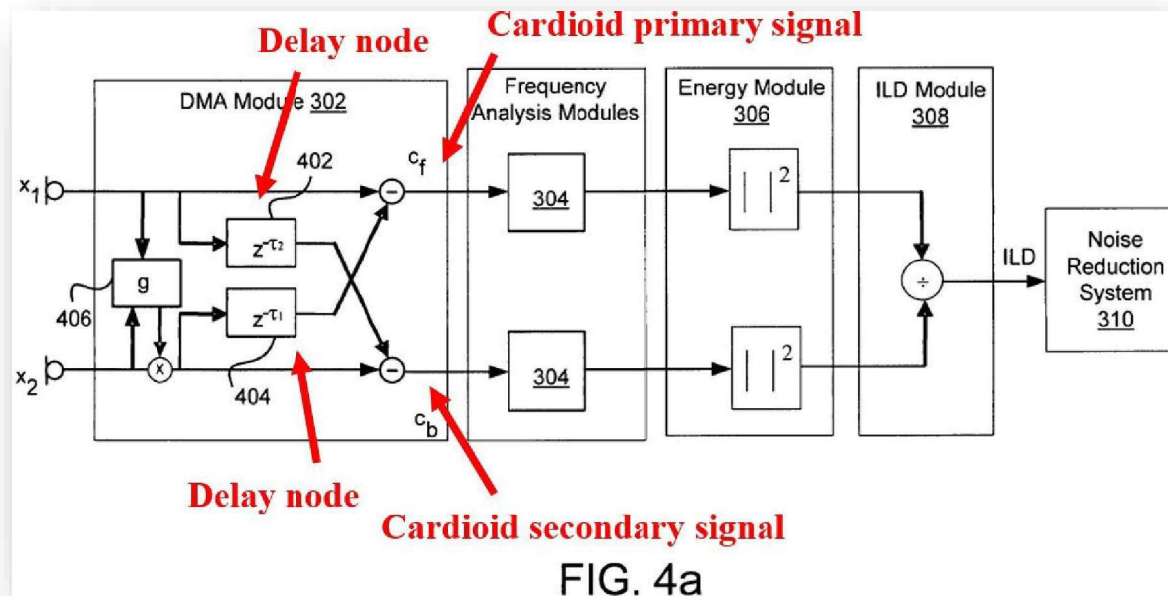


FIG. 4a

Ex. 1005, Fig. 4a

Prior Art

Avendano Annotated by Petitioner



Pet. at 10

Prior Art

Avendano Uses a Delay

Avendano's delay node, even if implemented as an all-pass filter, does not meet this requirement. Ex. 2005, ¶¶ 81-86.

Indeed, the only relationship Petitioner identifies is “a temporal relationship for speech between the first and second signals.” Pet., 26. Petitioner describes this relationship in nearly identical terms to the '213 Patent's delay filter:

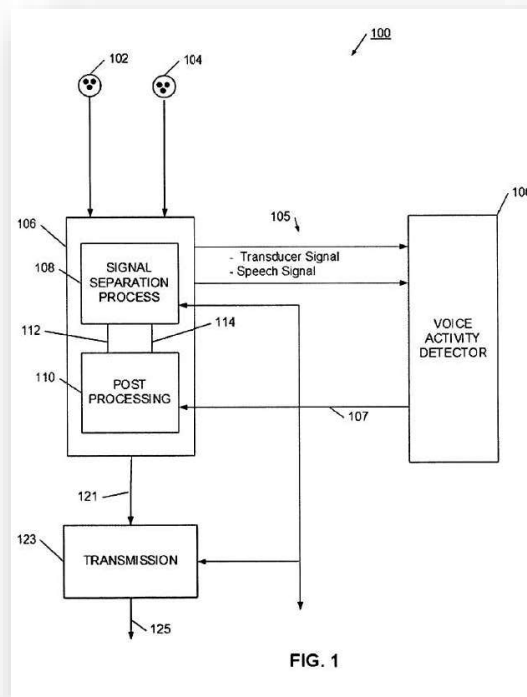
In Avendano, “due to a space difference between the microphones,” and because the speed of sound propagation in air is widely known to be approximately 330 m/s, there will be a “difference in times of arrival of the signal from a speech source to the microphones.”

Id.; Ex. 2005, ¶ 82.

Paper 29 at 25

Prior Art

Visser



Ex. 1006, Fig. 1

Prior Art

Visser Annotated by Petitioner

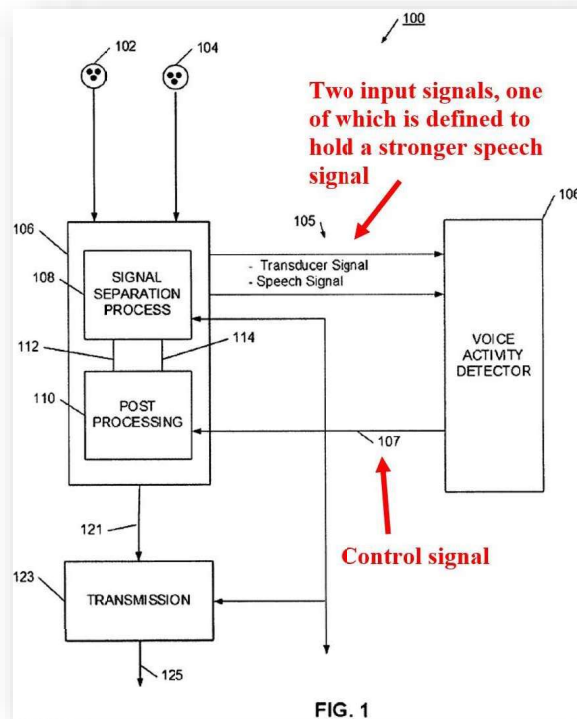
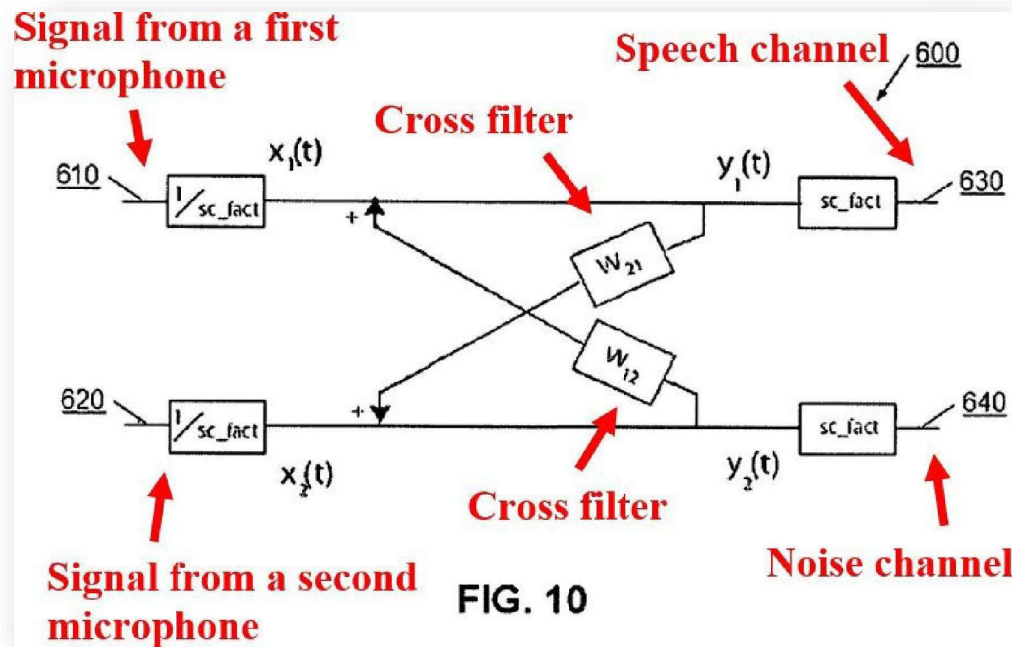


FIG. 1

Pet. at 12

Prior Art

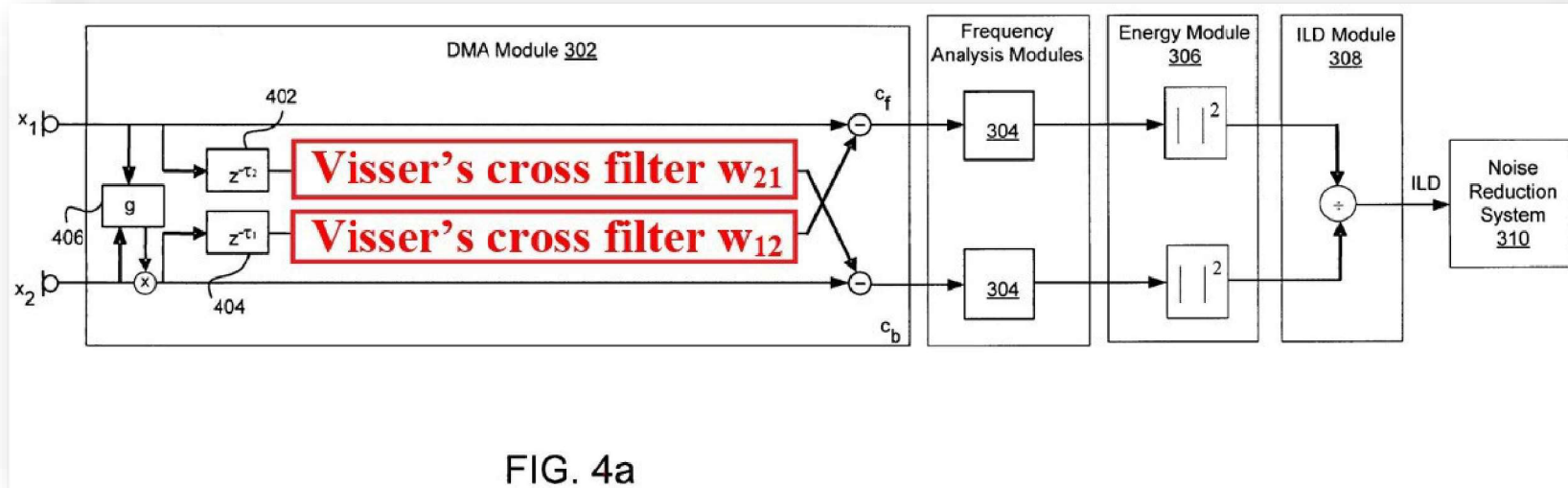
Visser Annotated by Petitioner



Pet. at 14

Petitioner's Combination

Avendano/Visser Combination Annotated by Petitioner



Pet. at 20

Petitioner's Combination

The Combination Does Not Include Visser's VAD

13 Just to be clear, I'm looking for
14 places where the voice activity
15 detection that's in Visser might have
16 been used to support my arguments and
17 I don't believe I explicitly rely on
18 the VAD in Visser to meet the
19 limitations where that's required. So
20 I think without wasting a lot more
21 time leafing through, I think the
22 answer to your question is the
23 components of Visser that I rely upon
24 for my opinions are associated with
25 Figure 10 and the structures within

Page 61

1 KENNY
2 Figure 10, and the description of what
3 goes on inside of the cross filters
4 and components around them.

Paper 29 at 28
Ex. 2004, 60:13-61:4

14 Other than the cross filters and
15 the description of what could be within
16 the cross filters and how the cross
17 filters may be connected, do you rely on
18 any other components of Visser for your
19 opinion?
20 MS. WANG: Objection to form.
21 THE WITNESS: I don't think
22 there's any explicit references to
23 other elements of the entire system of
24 Visser in support of my opinions.

Paper 29 at 28
Ex. 2004, 61:14-25

Petitioner's Combination

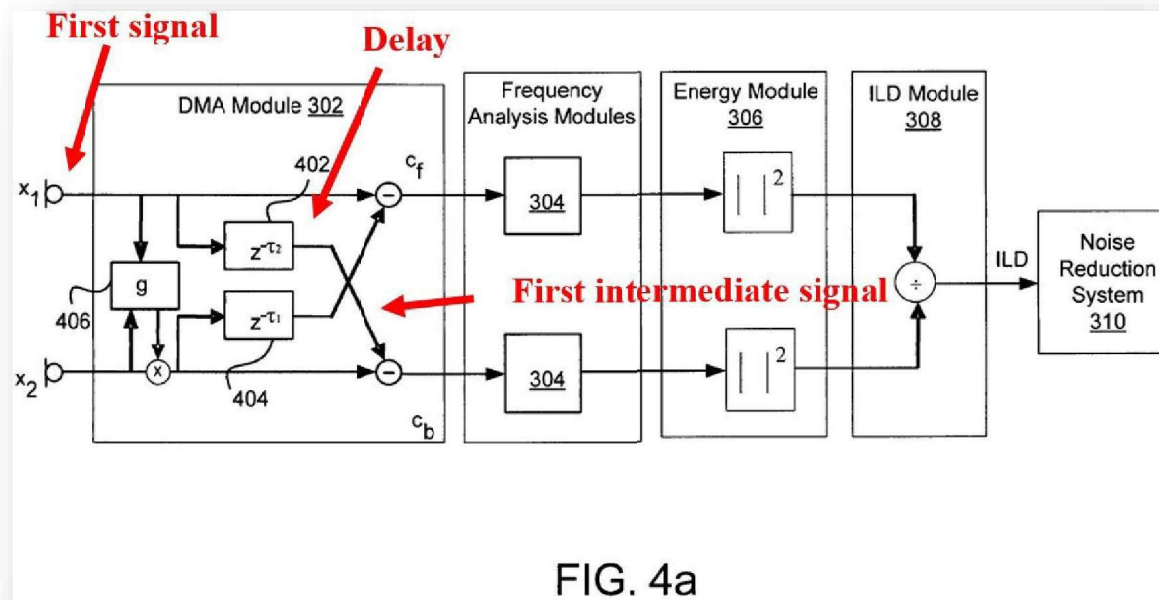
Visser's VAD Is Necessary

For purposes of explanation, the speech separation process will be described as an independent component analysis (ICA) process. Generally, the ICA module is not able to perform its main separation function in any time interval when the desired speaker is not speaking, and therefore may be turned off. This “on” and “off” state can be monitored and controlled by the voice activity detection module 185 based on comparing energy content between input channels or desired speaker a priori knowledge such as specific spectral signatures. By turning the ICA off when desired speech is not present, the ICA filters do not inappropriately adapt, thereby enabling adaptation only when such adaptation will be able to achieve a separation improvement. Controlling adaptation of ICA filters allows the ICA process to achieve and maintain good separation quality even after prolonged periods of desired speaker silence and avoid algorithm singularities due to unfruitful separation efforts for addressing situations the ICA stage cannot solve. Various ICA algorithms exhibit dif-

Paper 29 at 27-28
Ex. 1006, 61:14-25

Petitioner's Combination

Avendano Annotated by Petitioner (Claim 17)



Pet. at 36

Petitioner's Combination

Avendano/Visser Annotated by Petitioner (Claim 17)

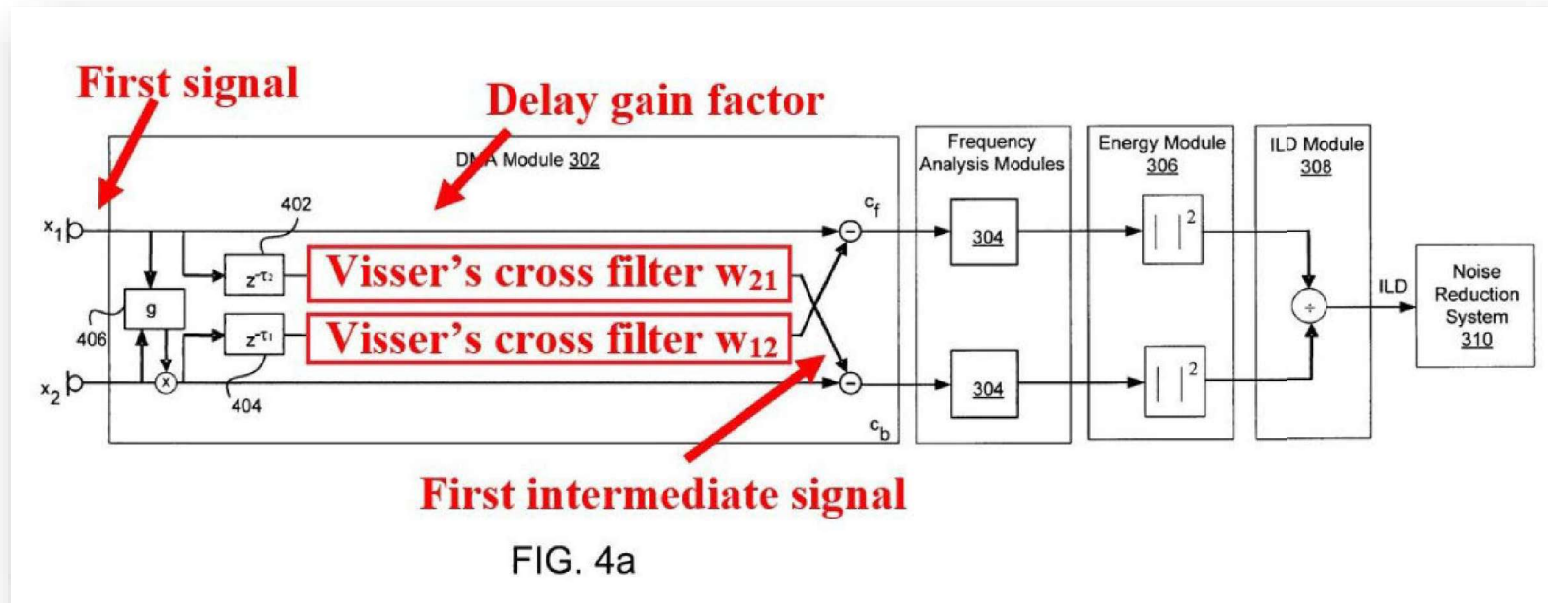
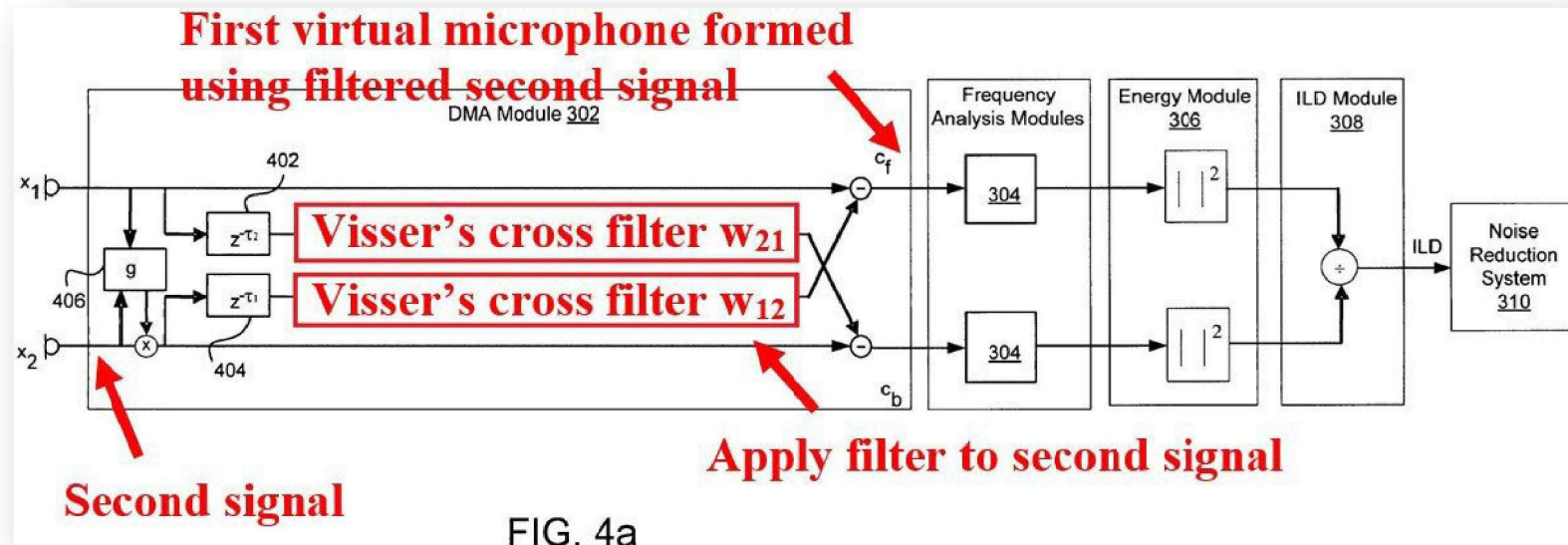


FIG. 4a

Pet. at 37

Petitioner's Combination

Avendano/Visser Combination Annotated by Petitioner (Claim 19)



Pet. at 47