US010535362B2

(12) **United States Patent**
Bryan et al.

(10) **Patent No.: US 10,535,362 B2**
(45) **Date of Patent: Jan. 14, 2020**

(54) **SPEECH ENHANCEMENT FOR AN ELECTRONIC DEVICE**

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(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **15/909,513**

(22) Filed: **Mar. 1, 2018**

(65) **Prior Publication Data**

US 2019/0272842 A1 Sep. 5, 2019

(51) **Int. Cl.**

G10L 21/0232	(2013.01)
G10L 21/028	(2013.01)
H04R 3/00	(2006.01)
G10L 21/02	(2013.01)
G10L 21/0216	(2013.01)

(52) **U.S. Cl.**

CPC ***G10L 21/0232*** (2013.01); ***G10L 21/0205*** (2013.01); ***G10L 21/028*** (2013.01); ***H04R 3/005*** (2013.01); *G10L 2021/02166* (2013.01)

(58) **Field of Classification Search**

None

See application file for complete search history.

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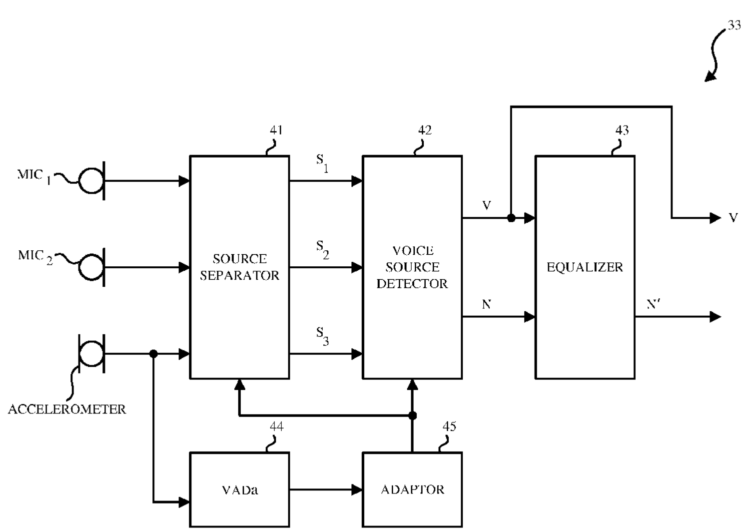
(57) **ABSTRACT**

Signals are received from audio pickup channels that contain signals from multiple sound sources. The audio pickup channels may include one or more microphones and one or more accelerometers. Signals representative of multiple sound sources are generated using a blind source separation algorithm. It is then determined which of those signals is deemed to be a voice signal and which is deemed to be a noise signal. The output noise signal may be scaled to match a level of the output voice signal, and a clean speech signal is generated based on the output voice signal and the scaled noise signal. Other aspects are described.

18 Claims, 8 Drawing Sheets



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