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- (54) **SPEECH ENHANCEMENT FOR AN ELECTRONIC DEVICE**
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None
See application file for complete search history.

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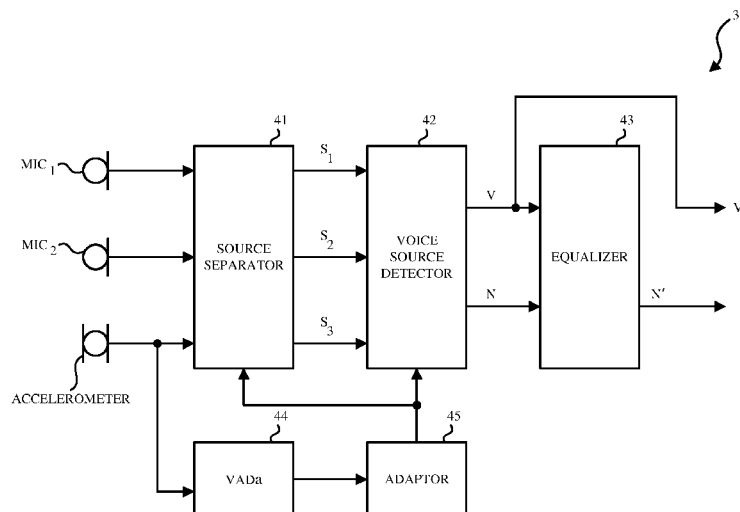
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(57) ABSTRACT

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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