Title (en)

Circuit and method for adaptive noise suppression

Title (de)

Schaltung und Verfahren zur adaptiven Geräuschunterdrückung

Title (fr)

Circuit et méthode pour la suppression adaptive du bruit

Publication

EP 1154674 A3 20070321 (DE)

Application

EP 01810057 A 20010122

Priority

CH 2042000 A 20000202

Abstract (en)

[origin: US6928171B2] The circuit for adaptive suppression of noise is a component part of a digital-hearing aid, consists of two microphones (1, 2), two AD-converters (3, 4), two compensating filters (5, 6), two retarding elements (7, 8), two subtractors (9, 10), a processing unit (11), a DA-converter (13), an earphone (15) as well as the two filters (17, 18). The method for adaptive suppression of noise can be implemented with the indicated circuit. The two microphones (1, 2), provide two differing electric signals (d₁(t), d₂(t)), which are digitalized in the two AD-converters (3, 4) and pre-processed together with the two fixed compensation filters (5, 6). Downstream the compensation filters are arranged the two filters (17, 18) symmetrically crosswise in a forward direction and having adaptive filter coefficients (w₁, w₂). The filter coefficients (w₁, w₂) are calculated by a stochastic gradient procedure and updated in real time while minimizing a quadratic cost function consisting of cross-correlation terms. As a result of this, spectral differences of the input signals are selectively amplified. With a suitable positioning of the microphones (1, 2) or selection of the directional characteristics, the signal to noise ratio of output signals (s₁, s₂) compared to that of the individual microphone signals (d₁(t), d₂(t)) can be significantly increased. Preferably, one of the two improved output signals (s₁, s₂) within one of the processing units (11, 12) is subjected to the usual processing specific to hearing aids, sent to one of the DA-converters (13, 14) and acoustically output once again through one of the earphones (15, 16). Four additional cross-over element filters (19-22) carry out a signal-dependent transformation of the input and output signals (y₁, y₂; s₁, s₂), and solely the transformed signals are utilized for the updating of the filter coefficients (w₁, w₂). This makes possible a rapidly reacting, and nonetheless calculation-efficient updating of the filter coefficients (w₁, w₂), and in contrast to other methods only causes minimal audible distortions.

IPC 8 full level

H04R 25/00 (2006.01)

CPC (source: EP US)

H04R 25/505 (2013.01 - EP US)

Citation (search report)

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