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

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Application

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Priority

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Abstract (en)

[origin: CA2332092A1] The circuit for the adaptive suppression of noise is a component part of a digital hearing aid, consisting 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 the adaptive suppression of noise can be implemented with the indicated circuit. The two microphones (1, 2), dependen t on their spatial arrangement or their directional characteristics and dependent on the location of the acoustic signal sources, 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). Following subsequently are the two filters (17, 18) arranged symmetrically crosswise in forward directi on with the adaptive filter coefficients (w1, w2). The filter coefficients (w1, w2) are calculated by means of 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, therefore the signal to noise ratio of output signals (s1, s2) in comparison with that of the individual microphone signals (d1(t), d2(t)) can be significantly increased. In preference, one of the two improved output signa ls (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). In the case of the invention presented here, 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 filt er 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 minimally audible distortions. (Figure 2)

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