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Datasheet for the decision of 25 October 2022

Case Number: T 2530/19 - 3.5.03

Application Number: 10752598.2

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IPC: H03G3/30, H03G3/32, H04R25/00

Language of the proceedings: EN

Title of invention:
Method and system for providing hearing assistance to a user

Patent Proprietor:
Sonova AG

Opponent:
Oticon A/S / Widex A/S / GN Hearing A/S

Headword:
Hearing aid's gain model/SONOVA

Relevant legal provisions:
EPC Art. 56, 83

Keyword:
Inventive step - main request and first auxiliary request (no), second auxiliary request (yes)
Sufficient disclosure - second auxiliary request (yes)

Boards of Appeal of the
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Case Number: T 2530/19 - 3.5.03

D E C I S I O N
of Technical Board of Appeal 3.5.03
of 25 October 2022

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Decision under appeal: **Interlocutory decision of the Opposition Division of the European Patent Office posted on 3 July 2019 concerning maintenance of the European Patent No. 2617127 in amended form.**

Composition of the Board:

Chair K. Bengi-Akyürek
Members: K. Schenkel
C. Heath

Summary of Facts and Submissions

I. The appeals by the patent proprietor and the opponent lie from the interlocutory decision of the opposition division maintaining the present European patent in amended form on the basis of a "first auxiliary request" filed with a letter of 5 March 2018.

II. In this decision, reference is made to the following prior-art documents:

E1: EP 1 819 195 A2;
E2: EP 1 729 410 A1;
E8: EP 1 748 677 A2;
E9: DE 10 2005 061 000 A1.

III. Oral proceedings were held before the board on 25 October 2022 by videoconference.

The parties' final requests were as follows:

- The appellant/opponent requests that the decision under appeal be set aside and that the patent be revoked.

- The appellant/patentee requests that the appeal be dismissed (**main request**), in the alternative, that the patent be maintained on the basis of one of **four auxiliary requests** filed with the written reply to the opponent's statement of grounds of appeal.

At the end of the oral proceedings, the board's decision was announced.

IV. Claim 1 of the **main request** reads as follows (board's labelling):

"A method for providing hearing assistance to a user, comprising:

(a) capturing input audio signals by a microphone arrangement (42);

(b) estimating a speech level of the input audio signals and an ambient noise level of the input audio signals;

(c) applying a gain model to the input audio signals in order to transform the input audio signals into filtered audio signals;

(d) transmitting the filtered audio signals by a transmission unit (10) via a wireless audio link (34) to a receiver unit (14, 36, 38) comprising or being connected to means (18, 26) for stimulating the hearing of a user, the stimulating means being a loudspeaker worn at or in the user's ear; wherein the received audio signals are supplied from the receiver unit to the stimulating means, and

(e) stimulating the user's hearing by the stimulating means according to the audio signals supplied by the receiver unit,

(f) characterized in that for each respective ambient noise level the gain applied to the input audio signals varies as a function of the speech level, wherein the function varies according to the ambient noise level in such a manner that the ratio of the gain at low speech levels and at high speech levels changes as a function of the ambient noise level."

V. Claim 1 of the **first auxiliary request** differs from claim 1 of the main request in that feature (b) now reads as follows (board's underlining of added text):

(b) "estimating a speech level of the input audio signals, by an input speech level estimation unit (52) comprised in a transmission unit (10), and an ambient noise level of the input audio signals, by an ambient noise estimation unit(48) comprised in the transmission unit (10);".

VI. Claim 1 of the **second auxiliary request** differs from claim 1 of the first auxiliary request in that feature (f) now reads as follows (board's underlining of added text):

(f) "characterized in that for each respective ambient noise level the gain applied to the input audio signals varies as a function of the speech level, wherein the function varies according to the ambient noise level in such a manner that the ratio of the gain at low speech levels and at high speech levels changes as a function of the ambient noise level, wherein the gain model comprises a linear range in which the gain is constant irrespective of the speech level of the input audio signals and a compressive range which is adjacent to the linear range and in which compressive range the gain decreases with a given slope from the constant gain value of the linear range with increasing speech level of the input audio signals, wherein the boundary between the linear range and the compressive range is formed by a kneepoint, and wherein the position of the kneepoint is a function of the ambient noise level while the slope of the compressive range remains fixed so that the

<u>constant gain value of the linear range varies according to the position of the kneepoint</u>."

Independent claim 11 of the **second auxiliary request** is directed to a "system" adapted to perform the method steps of claim 1.

Reasons for the Decision

1. Main request - inventive step (Article 56 EPC)

1.1 The opposed patent relates to a system and a method for providing a hearing assistance, wherein an audio signal is captured by means of a microphone arrangement. More specifically, a speech level and an ambient-noise level of the captured signal are estimated, a "gain model" is applied to the audio signal that is transmitted wirelessly to a receiver unit and a user's hearing is stimulated accordingly. A typical application is disclosed to be a wireless system in which the voice of a teacher is captured and transmitted to children (page 1, lines 13 to 20 of the description as originally filed). According to the gain model, a gain is applied to the audio signal which varies as a function of the incoming signal's speech level ("input speech level"). In a disclosed embodiment, the gain remains constant until a certain speech level is reached (called "kneepoint") and reduced beyond the "kneepoint", thereby causing a dynamic compression of louder input signals (page 4, line 28 to page 5, line 4 as filed).

Furthermore, the ratio of the gain at low speech levels and at high speech levels varies according to the ambient-noise level, i.e. the function of the gain over the speech level varies dependent on the ambient-noise level. In an embodiment, the "kneepoint" is shifted towards lower speech levels with decreasing ambient-noise levels (page 5, lines 5 to 12 as filed). The technical effect of this is that the gain at *high* speech levels is more decreased compared to the gain at *low* speech levels at lower ambient noise (see page 8, lines 3 to 14 and Fig. 3). This is said to render the speech signal less susceptive to changes in speech level which may be caused by variations in the distance between the speaker's mouth and the microphone (page 5, lines 5 to 12).

1.2 Claim 1 of the **main request** includes the following limiting features (board's labelling):

A method for providing hearing assistance to a user, comprising:

(a) capturing input audio signals by a microphone arrangement;
(b) estimating a speech level of the input audio signals and an ambient noise level of the input audio signals;
(c) applying a gain model to the input audio signals in order to transform the input audio signals into filtered audio signals;
(d) transmitting the filtered audio signals by a transmission unit via a wireless audio link to a receiver unit comprising or being connected to means for stimulating the hearing of a user, the stimulating means being a loudspeaker worn at or in the user's ear; wherein the received audio signals are supplied from the receiver unit to the stimulating means, and

(e) stimulating the user's hearing by the stimulating means according to the audio signals supplied by the receiver unit,

(f) characterized in that for each respective ambient noise level the gain applied to the input audio signals varies as a function of the speech level, wherein the function varies according to the ambient noise level in such a manner that the ratio of the gain at low speech levels and at high speech levels changes as a function of the ambient noise level.

1.3 In the proprietor's favour, the board interprets the expression "for each respective ambient noise level" recited in feature (f) such that it encompasses "<u>practically relevant</u> ambient noise levels" for the purpose of the following assessment of inventive step. Hence, for *different* practically relevant noise levels *different* gains in dependence of the detected input speech levels are applied to the received audio signals. In addition, as to the second part of feature (f), the parties agreed that the ratio between the gain at *low* and at *high* speech levels need not vary solely in dependence of the "ambient-noise levels", i.e. the different "kneepoints", according to its wording. In view of the breadth of that wording, this ratio may, for a particular gain curve, also vary between its "linear range" (implying a constant gain value) and its "compressive range" (implying a decreasing gain value), as illustrated e.g. by Figure 3 of the opposed patent.

1.4 Document **E1**, which is considered to be the most promising starting point for the assessment of inventive step, discloses a method and a system for providing a hearing assistance including capturing audio signals by means of a microphone arrangement, analysing the audio signals to determine an "audio scheme", setting a gain applied to the audio signals according to the determined audio scheme, transmitting wirelessly the audio signals to a receiver unit and stimulating the user's hearing accordingly (cf. abstract). A typical, disclosed application relates to transmitting a teacher's voice to children that are listening (paragraph [0003]). In a disclosed embodiment, the system includes a "transmission unit" worn by a speaker and a "receiver unit" worn by a listener (paragraph [0023], Fig. 1).

As to **features (a) and (d)**, the "transmission unit" includes a "microphone arrangement" for capturing the speaker's voice and producing audio signals to which data signals are added and which are supplied to a wireless "FM transmitter" (paragraph [0026]).

As to **features (d) and (e)**, the "receiver unit" includes a "receiver 124" for receiving the audio signals and the data signals, a "DTMF decoder" for decoding the data/command signals, an "amplifier 126" followed by a "power amplifier 137" for amplifying the audio signals and a "loudspeaker" for stimulating the listener's hearing accordingly (paragraph [0033]).

As to **feature (b)**, the "radio transmission unit" includes a "voice energy estimator unit" for computing the total energy in the underlying voice spectrum, i.e. speech level, which information is used by a "voice judgement unit" to decide whether a voice signal is present or not, and a "surrounding noise level estimator unit" for estimating the surrounding noise level, i.e. ambient-noise level (paragraphs [0027] to [0029]). The information about the presence of voice and the surrounding noise level is included in the data transmitted to the "receiver unit" and decoded there in the "DTMF decoder" (paragraphs [0029] and [0031] and paragraph [0033], last sentence).

As to **features (c) and (f)**, if the "voice judgement unit" of the "transmission unit" decides that voice is present, it provides the value "Voice ON" and the gain of the "amplifier 126" of the "receiver unit" is set to a given value, whereas in the opposite case, the value "Voice OFF" is provided and the gain is set to a lower value, i.e. the gain varies as a function of the voice/speech level (paragraph [0036]).

As to **feature (f)**, in a certain embodiment, in the "Voice ON" case, which corresponds only to high speech levels, a "parameter update unit 129" of the "receiver unit" controls the "amplifier 126" depending on the surrounding noise level such that the "amplifier 126" applies an additional gain offset to the audio signal, i.e. the ratio of the gains for *low* and for *high* speech levels respectively changes as a function of the ambient-noise level (paragraph [0037], last two sentences, Fig. 6). Thus, in view of the breadth of feature (f) and in contrast to the opposition division's view (cf. Reasons 2.4.2.1 of the appealed decision), this embodiment falls well within the terms of that feature.

1.5 The method of claim 1 therefore differs from the method of E1 only in that the "gain model" is applied to the audio signal <u>prior</u> to its wireless transmission.

1.6 The patentee argued that E1 did not clearly and unambiguously disclose that the "additional gain offset" is only applied in the "Voice ON" event.

Furthermore, the additional gain offset would only be applied if the situation of *high* speech levels were to arise and could not be compared with the gain in the non-existing situation of *lower* speech levels which is not part of the diagram. Thus, there could not be a change of the gain ratio as required by feature (f) of claim 1.

1.7 The board is not convinced by these arguments. The last two sentences of paragraph [0037] in fact read as follows (board's underlining):

> *"According to alternative embodiments, the 'surrounding noise level' is estimated <u>only</u> or also <u>during "voice ON"</u>. <u>In these cases</u>, during "voice ON", the parameter update unit 129 controls the amplifier 126 depending on the "surrounding noise level" such that according to the definition stored in the EEPROM 130 the amplifier 126 applies an additional gain offset to the audio signals sent to the power amplifier 137."*

The first sentence mentions two cases, in the first of which <u>only</u> in the "Voice ON" situation with higher speech levels the surrounding noise level is estimated. The following sentence refers to *both* cases, i.e. also to the aforementioned one. The board agrees that the "extra gain" is applied in the "Voice ON" situation which by nature cannot coexist with the "Voice OFF" situation at the same time. However, the "Voice OFF" situation and the rules for calculating the gain in this situation do not disappear because of that. The gains in these two situations can therefore indeed be compared and a change of the gain ratio actually be established in accordance with feature (f).

1.8 The technical effect of shifting the application of the gain model to the transmitter side is merely that the overall processing requirements are shifted to the transmitter side.

1.9 The proprietor argued that the objective technical problem is to be seen in "increasing the speech intelligibility and the user's comfort". The board takes issue with that objective problem. This is because only the different *location* of where the gain model is eventually applied is acknowledged as distinguishing feature which however has no impact on the speech intelligibility or the user's comfort. The objective technical problem is rather to be seen in *reducing signal processing requirements at the receiver side in the hearing system of E1*.

1.10 Given the situation that all necessary information for applying the gain model, namely the estimation of the speech level and of the ambient-noise level, is already present on the transmitter side in the system of E1, the skilled person in the field of hearing aids, starting out from E1, would have readily applied the corresponding gain model right away on the *transmitter* side to reduce implementation complexity on the receiver side without having to exercise inventive skills.

The main request is therefore not allowable under Article 56 EPC.

2. First auxiliary request - inventive step (Article 56 EPC)

2.1 Claim 1 of the first auxiliary request differs from claim 1 of the main request only in that

- the speech level is estimated by an <u>input speech level estimation unit</u> in a transmission unit and

- the ambient noise level is estimated by an <u>ambient noise estimation unit</u> in the transmission unit.

Both features are disclosed in E1 (cf. "voice energy estimator unit 114" and "surrounding noise level estimator unit 117", paragraphs [0028] and [0029], Fig. 4).

2.2 Therefore, the added features cannot contribute to an inventive step and the first auxiliary request is not allowable under Article 56 EPC either.

3. Second auxiliary request - Articles 83 and 56 EPC

3.1 Sufficiency of disclosure (Article 83 EPC)

3.1.1 The opponent argued that there is no sufficient disclosure as to feature (b) and the "ratio" of the gains at "high speech levels" and "low speech levels" in feature (f). Speech was always contaminated by noise and vice versa and it was not trivial to obtain estimates for the speech level and the ambient-noise level. The opposed patent disclosed only speech and a noise estimation units but without any description how such an estimation should work and therefore failed to enable the skilled person to put the claimed subject-matter in practice. Likewise, the description of the embodiments did not disclose a ratio of the gain at low speech levels and at high speech levels or any indication of how to implement the ratios as a function of the ambient noise level. Finally, it was not disclosed what constitutes a *low* speech level and a *high* speech level. The boundary between low and high speech level could theoretically be set at an extreme end of the dynamic range of the user's hearing contravening the gist of the invention. In the embodiment disclosed in paragraph [0031] and Figure 3, the gain would decrease for speech levels above 73 dB irrespective of the kneepoint.

3.1.2 The board holds that it belongs to the common general knowledge of a skilled person in the field of hearing aids to estimate the speech and the noise level in a sound signal, for example by analysing its spectral components. A further detailed teaching is not necessary. The ratio of the gain at *lower* speech levels and *high* speech levels simply specifies the quotient between two gain values at two different speech levels and needs no further explanation. The respective speech levels are specified as lower and higher which is self-explanatory as well. It is further noted that Figures 2 and 3 with their corresponding description on page 8, lines 3 to 14 depict how the gain for lower speech levels can be increased compared to the gain at higher speech levels by shifting the kneepoint toward lower speech levels with decreasing ambient noise levels which results in the three curves in Figure 3, the upper one corresponding the lowest ambient noise level.

3.1.3 The board therefore concludes that the claimed invention is sufficiently disclosed (Article 83 EPC).

3.2 Inventive step (Article 56 EPC)

3.2.1 In claim 1 of the second auxiliary request, the gain model is now limited to

(g) a <u>linear range</u> in which the gain is constant irrespective of the speech level, and

(h) a <u>compressive range</u> which is adjacent to the linear range and in which the gain decreases with a given slope from the constant gain value of the linear range with increasing speech level,

(i) wherein the boundary between the <u>linear range</u> and the <u>compressive range</u> is formed by a <u>kneepoint</u>, wherein the position of the kneepoint is a <u>function</u> of the ambient-noise level,

(j) while the slope of the <u>compressive range</u> remains <u>fixed</u> so that the constant gain value of the linear range varies according to the position of the kneepoint.

Added features (g) to (j) indeed properly reflect that different "gain versus speech level" curves are derived from different "kneepoints" depending on distinct "ambient-noise levels", as also illustrated e.g. in Figures 2 and 3 of the opposed patent. Hence, at *high* speech levels, the gain decreases with an increasing speech level independently of the ambient-noise level, while, at *low* speech levels, the gain depends on the ambient-noise level but not on the speech level. Moreover, the transition point between these regions shifts dependent on the ambient-noise level. This apparently brings about the technical advantage that the "effect of the distance of the microphone to the mouth of the speaker on the sound level in quiet conditions and at lower speech levels is reduced" and that "the allowed maximal distance between the speaker and the microphone is increased" (see paragraph [0016] of the opposed patent).

3.2.2 Document E1 is silent as to the added features (g) to (j), which as a whole allow a specific user experience including a consistent compression of *higher* speech levels and a maximum gain at *lower* speech levels, while limiting the incurred noise in the amplified audio signal. Thus, the underlying objective technical problem may now be seen in *improving the signal processing in the hearing assistance of E1 at all possible speech levels*.

3.2.3 The person skilled in the field of hearing aids, starting from E1 and faced with the above objective problem, would have recognised that, in E1, the gain at higher speech levels ("Voice ON" region) is increased and that the additional gain dependent on the ambient-noise level is applied at *higher* speech levels and not at *lower* speech levels. Hence, the skilled person would readily notice that the gain model of E1 goes in exactly the opposite direction compared to that of present claim 1, thereby teaching away from the claimed solution.

3.2.4 In that context, the opponent referred to document **E8** (in particular paragraphs [0001] and [0011]) and argued that the document disclosed a gain curve with two essentially linear ranges and a kneepoint in between which is preferably placed below an input signal range. Further, document **E9** (in particular Fig. 2) disclosed shifting the kneepoint based on the ambient-noise level which is said to be reflected by a situation classified by a "classifier" based on the respective hearing situation.

3.2.5 As to prior-art document **E8**, it is apparent to the board that the kneepoint is set statically but is not shifted in dependence on the ambient noise level (cf. paragraph [0011]. Moreover, document **E9** discloses two gain curves, each with three linear regions separated by two kneepoints, which are moved up or down, depending on the present hearing situation. A shift of the kneepoints towards higher or lower signal levels is however not disclosed, nor is it disclosed that the hearing situation reflects the respective ambient-noise level.

3.2.6 The board therefore concludes that the skilled person, starting out from E1, considering E8, E9 or common general knowledge and faced with the above objective problem would indeed not arrive at a method with all the features of claim 1 without exercising inventive skills (Article 56 EPC).

3.2.7 In view of the above, claim 1 and also independent claim 11 of the second auxiliary request are allowable under Article 56 EPC.

3.3 Given that no other objections were invoked by the opponent or the board, the claims of the second auxiliary request comply with all requirements of the EPC.

Order

For these reasons it is decided that:

1. The decision under appeal is set aside.

2. The case is remitted to the opposition division with the order to maintain the patent in the following version:

   - Claims 1 to 11 of the second auxiliary request as filed with letter dated 5 March 2020;

   - Description, pages 2 and 4 to 6 of the patent specification, page 3 as filed during the oral proceedings before the Board;

   - Drawings, figures 1 to 6 of the patent specification.

The Registrar: The Chair:

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B. Brückner K. Bengi-Akyürek

Decision electronically authenticated