FEDERAL SUPREME COURT

IN THE NAME OF THE PEOPLE

JUDGMENT

X ZR 98/23

Announced on:
July 15, 2025 Dreixler
Court clerk as registrar
of the registry

in the patent nullity case

Reference work:	Yes
BGHZ:	No
BGHR:	yes
JNEU:	Yes

Late reverberation

Patent Act (PatG) § 81, § 99 (1); Code of Civil Procedure (ZPO) § 263

a) A change of party and thus also the joinder of another party as plaintiff can be declared as long as the legal dispute is pending.

b) An effective declaration of accession does not lose its effect if the first plaintiff withdraws the complaint and the declaration of withdrawal is served on the defendant before the declaration of accession.

EPÜ Art. 87

There is a rebuttable but strong presumption in favor of the right to claim priority (confirmation by the EPO, decision of October 10, 2023 - G 1/22, para. 86, para. 101 et seq. and para. 122 - right to priority; Federal Supreme Court (BGH), judgment of November 28, 2023 - X ZR 83/21, GRUR 2024, 374 para. 110 et seq. - Sorafenib-Tosylat; Judgment of January 9, 2024 - X ZR 74/21, GRUR 2024, 603 para. 67 et seq. - Happy Bit).

Federal Supreme Court, judgment of July 15, 2025 - X ZR 98/23 - Federal Patent Court

ECLI:DE:BGH:2025:150725UXZR98.23.0

The X Civil Senate of the Federal Supreme Court, following oral proceedings on July 15, 2025, presided over by Judge Dr. Bacher, Judge Hoffmann, Judge Dr. Kober-Dehm, Judge Dr. Rensen, and Judge Dr. von Pückler,

ruled as follows:

> The appeals against the judgment of the 5th Senate (Nullity Senate) of the Federal Patent Court of April 17, 2023, are dismissed.
>
> Two-thirds of the costs of the appeal proceedings shall be borne by the plaintiff in 2 and one-third by the defendant.

By law

Facts of the case:

1 The defendant is the owner of European patent 1 565 036 (the patent in suit), which was granted with effect for the Federal Republic of Germany and was filed on February 4, 2005, claiming two US priority dates of February 12, 2004, and April 1, 2004, and relates to the processing of audio signals.

2 Claim 1, to which six claims refer, reads as follows in the procedural language:

> A method of audio processing for synthesizing an auditory scene, comprising: processing (702) at least one input channel (312), using an auditory filter bank block (702), to generate two or more processed input signals (704);
> filtering (720) the at least one input channel (312), using a filter (720) that models late reverberation (LR), to generate corresponding two or more LR-filtered diffuse signals (722); and
> for each of the two or more processed input signals and each of the corresponding two or more diffuse signals, combining (714) one of the two or more LR-filtered diffuse signals with a corresponding one of the two or more processed input signals to generate one of a plurality of output channels (324) for the auditory scene.

3 Claim 8 and claim 9, to which a further claim refers, protect devices that are suitable for applying this method and have similar but differing features in detail.

4 In its nullity action filed on October 12, 2020, the first plaintiff asserted that the subject matter of the contested patent went beyond the content of the original documents and was not patentable. In a writ dated October 28, 2020, the second plaintiff declared its intention to join the proceedings as an additional plaintiff and asserted the same grounds for nullity.

5 The first plaintiff consented to the joinder in a writ dated October 29, 2020. In a writ dated October 30, 2020, it withdrew its complaint. In a writ dated the same day, the defendant consented to the withdrawal and stated that it would not submit a request for costs.

6 The statement of claim was served on the defendant on November 5, 2020, the withdrawal of the action on November 9, 2020, the declaration of intervention by the second plaintiff and the consent of the first plaintiff on November 12, 2020.

7 The defendant requested that the complaint be dismissed as inadmissible because the plaintiff's intervention was invalid or, in any case, irrelevant. In addition, it defended the contested patent in the granted version and, in the alternative, in 23 amended versions.

8 The Patent Court declared the contested patent invalid insofar as it went beyond the version according to auxiliary request 2A and dismissed the complaint in all other respects.

9 This is the subject of the appeals lodged by both parties, which are each pursuing their requests from the first instance.

Reasons for the decision:

10 Both appeals are admissible but do not contain reasons.

11 A. The complaint is admissible.

12 I. The Patent Court was right to consider the change of parties brought about by the declaration of accession of the second plaintiff and the declaration of withdrawal of the first plaintiff to be effective.

13 1. Contrary to the defendant's opinion, the intervention of the second plaintiff is not invalid because this declaration was only served on it after the withdrawal declaration of the first plaintiff.

14 a) A change of party and thus also the accession of a further party as plaintiff can in any case be declared as long as the legal dispute is pending.

15 This condition was met at the time of the accession on October 29, 2020, because the complaint brought by the first plaintiff had already been filed on that date and the declaration of withdrawal was not received by the court until one day later.

16 b) With the receipt of the declaration of withdrawal on October 30, 2020, the pendency of the complaint brought by the first plaintiff (which had not yet been served at that time) ceased to exist. However, this did not render the declaration of accession ineffective.

17 c) The isolated service of the complaint and the withdrawal notice of the first plaintiff prior to the service of the declaration of accession of the second plaintiff did not render this declaration ineffective either.

18 The effect of filing a complaint - the pendency of the proceedings - occurs regardless of the defendant's knowledge (see Becker-Eberhard in Münchener Kommentar zur ZPO, 7th ed. 2025, § 253 para. 14). The same applies to a declaration by which a third party joins pending proceedings as an additional party.

Such a declaration has the same substantive objective as the original complaint. Its effectiveness is therefore subject to the same principles.

19 Contrary to the defendant's opinion, effective joinder was possible even before the first complaint was served. This applies regardless of when a procedural legal relationship between the plaintiff and the defendant arises in the patent nullity proceedings. In any case, a legal relationship with the court arises as soon as the complaint is filed and thus becomes pending. This forms a sufficient basis for amending a complaint before it is served or for making other procedural declarations. The declarations that are permissible thereafter also include joining as an additional plaintiff. A subsequent withdrawal of the complaint and the service of this declaration do not invalidate such a declaration.

20 2. The decision of the Patent Court that the resulting change of plaintiff is relevant is not subject to review in the appellate instance pursuant to § 268 Code of Civil Procedure (ZPO).

21 a) In patent nullity proceedings, a change of plaintiff is to be treated as an amendment to the claim. Its admissibility is generally governed by the general rules of civil procedure law (Federal Supreme Court (BGH), judgment of October 14, 2014 - X ZR 35/11, GRUR 2015, 159 para. 10 – Zugriffsrechte; judgment of June 28, 1994 - X ZR 44/93, GRUR 1996, 865, juris para. 13 - Parteiwechsel).

22 b) In the case in dispute, it is not necessary to decide whether an action for patent invalidity becomes pending upon filing or, in accordance with paragraph 253(1) of the German Code of Civil Procedure (ZPO), only upon service.

23 Before the case becomes pending, an amendment to the complaint is admissible without further ado; the amended complaint replaces the original complaint that has not yet been served (Bacher in BeckOK Code of Civil Procedure (ZPO), as of March 1, 2025, Section 263 para. 7).

24 After the action has become pending, a change of plaintiff is permissible if the defendant agrees or the court considers the amendment to be relevant (BGH, judgment of June 28, 1994 - X ZR 44/93, GRUR 1996, 865, juris para. 13). If the court of first instance has affirmed the relevance, this decision is not contestable pursuant to § 268 ZPO (Federal Supreme Court, judgment of January 20, 1987 - X ZR 70/84, GRUR 1987, 351, juris para. 11 - Mauerkasten II).

25 In the case in dispute, the change of parties effected by the declarations is therefore admissible simply because the Patent Court considered it relevant.

26 II. The expiry of the patent in suit also does not preclude the admissibility of the complaint.

27 1. After a patent has expired, a nullity action is only admissible to the extent that the plaintiff has an interest worthy of protection in the revocation.

28 According to the case law of the Senate, such a need for legal protection exists if the plaintiff has reason to fear that he could be sued on the basis of the patent for actions taken in the period prior to its expiry. It is not necessary for the plaintiff to have been sued or warned for infringement of the patent (BGH, decision of March 12, 1981 - X ZB 16/80, juris para. 14 f. - Anzeigegerät). It is sufficient if he can demonstrate that he has reason to fear that he may still be subject to claims for past actions even after the expiry of the term of protection (BGH, judgment of June 20, 2023 - X ZR 31/21, GRUR 2023, 1178 para. 15 - Leistungsüberwachungsgerät). However, there is no need for legal protection for a complaint against an expired patent if the patent owner has waived all claims under the patent (BGH, judgment of September 9, 2010 - Xa ZR 14/10, GRUR 2010, 1084 para. 10 – Windenergiekonverter).

29 2. In the case in dispute, the second plaintiff (hereinafter: plaintiff) therefore has an interest worthy of protection in the declaration of invalidity.

30 According to the plaintiff's undisputed submission, the defendant has issued a warning to another customer of the plaintiff based on the disputed patent. This gives the plaintiff cause for concern that it may be held liable for any claims the defendant may have against the customer. This provides sufficient reasons for legal protection.

31 B. Both appeals do not have reasons.

32 I. The patent in suit relates to the processing of audio signals.

33 1. According to the description of the contested patent, the human brain can use the differences in time and level at which an audio signal reaches the two ears to give the listener the spatial perception that the signal originates from a source located at a specific position relative to them (paragraph 3).

34 a) This processing capability can be used to create auditory scenes in which audio signals from one or more sources are specifically modified to give the impression that the various audio sources are located at different positions relative to the listener (paragraph 4).

35 A conceivable application for this would be video conferences in which the video images of the participants are arranged around a conference table. Through appropriate modifications, the server could process the conventionally used mono signal and split it into two signals in such a way that a spatial impression is created (paragraph 8 f.).

36 b) Such a system requires a high transmission bandwidth, as the server must transmit a left and a right audio signal to each conference participant (paragraph 9).

37 US patent applications 09/848877 (published as US 2003/0026441, NK10) and 10/045458 (published as US 2003/0035553, G2) solved this problem by having each participant's PC receive only a single mono signal in which parameters for auditory scenes were embedded. These parameters could include, for example, information about the level or time difference between the individual channels (interchannel level difference, ICLD; inter-channel time delay, ICTD, paragraph 10).

38 The approach proposed in NK10 is based on the assumption that frequency subbands in which the energy of a source signal dominates that of all other source signals can be assigned exclusively to that specific source. Accordingly, different sets of parameters would be defined for different frequency bands (paragraph 11). The term BCC (binaural cue coding) is used for this type of coding. The same technique is referred to as PCSC (perceptual coding of spatial cues) in NK10 and G2 (paragraph 12).

39 G2 uses BCC to generate a combined signal in which different sets of parameters for auditory scenes are embedded in such a way that it can be processed by both BCC-based decoders and conventional receivers. For the latter, the additional information is transparent (paragraph 13). Such a mono signal requires only about 50 to 80 percent of the bandwidth needed for a stereo signal with two channels (paragraph 14).

40 c) The wider the audio source, the lower the coherence of a binaural signal. In an orchestra, it is therefore typically lower than in a soloist (paragraph 15).

41 Signals generated using the method described in NK10 and G2 have a coherence value close to the maximum possible value of 1. This leads to errors in the sound image, such as an overly "dry" acoustic impression, if the coherence value of the original signal is lower (paragraph 16).

42 In addition, the method according to NK10 and G2 is susceptible to inaccurate estimates of the BCC parameters. Especially when listening through headphones, this can create the impression that an object in a fixed position is moving randomly (paragraph 19).

43 US patent application 10/155437 (published as US 2003/0219130, NK9) therefore proposes incorporating BCC parameters based on the coherence of the input signals (paragraph 18). This technique tends to work better at high frequencies than at low frequencies (paragraph 20).

44 2. Against this background, the patent in suit addresses the technical problem of providing a method that avoids the disadvantages described above with as little effort as possible.

45 3. To solve this problem, the contested patent proposes a method in claim 1 whose features can be broken down as follows:

46

1.1	A method of audio processing for synthesizing an auditory scene, comprising	Verfahren des Audio-Verarbeitens zum Synthetisieren einer auditiven Szene, aufweisend
1.2	processing (702) at least one input channel (312), using an auditory filter bank block (702), to generate two or more processed input signals (704);	Verarbeiten (702) von mindestens einem Eingangskanal (312), unter Verwendung eines Auditiv-Filterbank-Blocks (702), um zwei oder mehr verarbeitete Eingangssignale (704) zu erzeugen;
1.3	filtering (720) the at least one input channel (312), using a filter (720) that models late reverberation (LR), to generate corresponding two or more LR-filtered diffuse signals (722); and	Filtern (720) des mindestens einen Eingangskanals (312), unter Verwendung eines Filters (720), der späten Widerhall (LR) modelliert, um entsprechende zwei oder mehr LR-gefilterte, diffuse Signale (722) zu erzeugen; und
1.4	for each of the two or more processed input signals and each of the corresponding two or more diffuse signals, combining (714) one of the two or more LR-filtered diffuse signals with a corresponding one of the two or more processed input signals to generate one of a plurality of output channels (324) for the auditory scene.	für jedes der zwei oder mehr verarbeiteten Eingangssignale und jedes der entsprechenden zwei oder mehr diffusen Signale, Kombinieren (714) von einem der zwei oder mehr LR-gefilterten, diffusen Signale mit einem entsprechenden einen der zwei oder mehr verarbeiteten Eingangssignale, um einen von einer Mehrzahl von Ausgangskanälen (324) für die auditive Szene zu erzeugen.

47 4. The differences between claims 1, 8, and 9 are summarized in the following table.

48

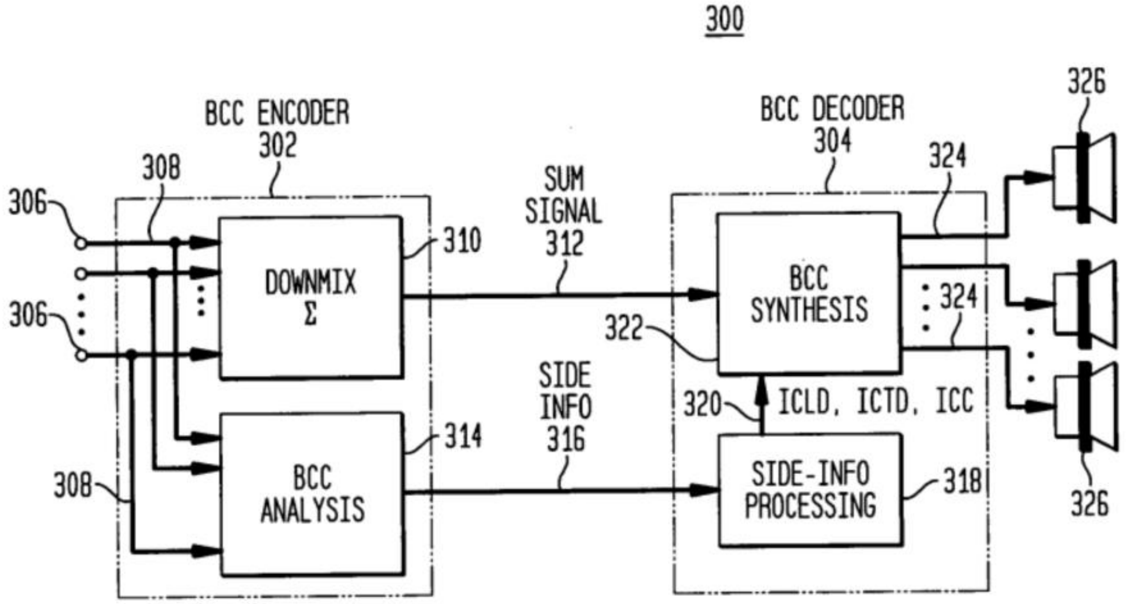
1.1	A method of audio processing for synthesizing an auditory scene, comprising	8.1	Apparatus (322) for audio processing including synthesizing an auditory scene, comprising:	9.1	Apparatus (322) for audio processing including synthesizing an auditory scene, comprising:
1.2	processing (702) at least one input channel (312), using an auditory filter bank block (702) to generate two or more processed input signals (704);	8.2	means (702) for processing at least one input channel (312) to generate two or more processed input signals (704);	9.2	a configuration of at least one time domain (TD) to frequency domain (FD) converter (702) and a plurality of filters (720) that model late reverberation (LR), the configuration adapted to generate two or more processed FD input signals (704) and corresponding two or more LR-filtered diffuse FD signals (722) from the at least one TD input channel (312);
1.3	filtering (720) the at least one input channel (312), using a filter (720) that models late reverberation (LR), to generate corresponding two or more LR-filtered diffuse signals (722); and	8.3	means (720) for filtering the at least one input channel (312), using a filter that models late reverberation (LR) to generate corresponding two or more LR-filtered diffuse signals (722); and		
1.4	for each of the two or more processed input signals and each of the corresponding two or more diffuse signals, combining (714) one of the two or more LR-filtered diffuse signals with a corresponding one of the two or more processed input signals to generate one of a plurality of output channels (324) for the auditory scene;	8.4	means (714) for combining, for each of the two or more processed input signals and each of the corresponding two or more diffuse signals, one of the two or more LR-filtered diffuse signals with a corresponding one of the two or more processed input signals to generate one of a plurality of output channels (324) for the auditory scene.	9.3	two or more combiners (714), each being adapted to combine one of the two or more LR-filtered diffuse FD signals (730) with a corresponding one of the two or more processed FD input signals (712) to generate a plurality of synthesized FD signals (716); and
				9.4	two or more frequency domain to time domain (FD-TD) converters (718), each adapted to convert one of the synthesized FD signals (716) into one of a plurality of TD output channels (324) for the auditory scene.

49 5. Some features require explanation.

50 a) By synthesizing an auditory scene in the sense of feature 1.1, an audio signal is generated by specifically modifying an input channel to convey a spatial perception.

51 aa) The basic features of such a synthesis process using BCC technology are shown in Figure 3 below.

FIG. 3



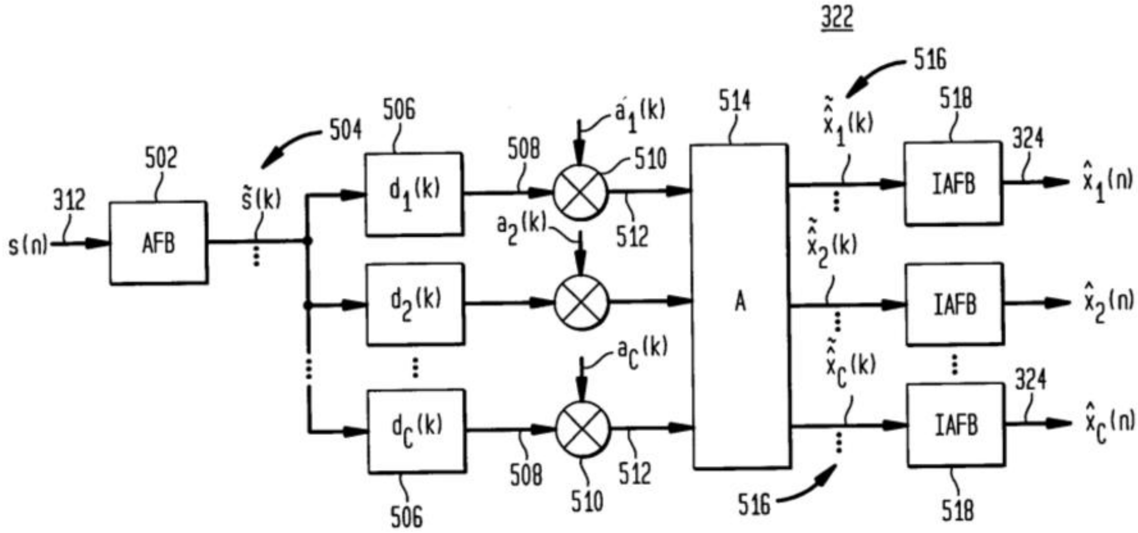
52 The BCC encoder (302) combines several input channels (308), each of which may originate from a microphone (306), by summing them into a single channel (312). In addition, BCC analysis generates additional information (316) that may contain details on the level and time difference as well as on the correlation (ICLD, ICTD, ICC) between the channels to be output (paragraph 24 f.).

53

Based on the sum signal (312) and the additional information (316), a BCC synthesizer (322) belonging to the BCC decoder (304) can generate several output channels (324) (paragraph 26).

54 bb) The operation of the BCC synthesizer (322) shown in Figure 3 is schematically illustrated in Figure 5 below.

FIG. 5



55 The combined input channel (312) is converted by an auditory filter bank block (502) into multiple copies of a signal (504) from the frequency domain (paragraph 36). Each of these copies is delayed by a delay block (506) based on the time difference information (ICTD) contained in the additional information and scaled by a multiplier (510) based on the level difference information (ICLD) (paragraph 37). The resulting signals (512) are fed to a coherence processor (514), which synthesizes a frequency domain signal (516) for each output channel (324) based on coherence information from the additional information. These signals are converted into time domain output channels (324) by inverse auditory filter bank blocks (paragraph 38).

56

b) Features 1.1 and 1.2 do not specify this procedure in all details.

57 aa) The requirement defined in feature 1.1, according to which the audio processing serves to synthesize an auditory scene, implies that the method must comprise a synthesis process of the type described above, in which an input channel must be converted into at least two different input signals on the basis of additional information.

58 Claim 1 does not necessarily specify where this additional information comes from.

59 bb) Feature 1.2 further specifies that at least one input channel is processed during synthesis and that at least two processed input signals are generated from this channel.

60 Contrary to the defendant's view, this does not preclude more than one input channel being processed into at least two processed input signals in each case. Nor does it preclude several of the processed input signals being recombined in a subsequent step.

61 cc) Feature 1.2 further stipulates that the input signal (312) is processed using an auditory filter bank block.

62 (1) As the Patent Court correctly assumed at the outset, such a block must consist of a plurality (bank) of filters arranged in one or more dimensions.

63 (2) There is no need to decide whether feature 1.2 – as assumed by the Patent Court – does not provide any further specifications for the design of the auditory filter bank block or whether the filtering – as the defendant believes – must necessarily lead to a division into several frequency bands.

64 In this respect, the Patent Court correctly assumed that the contested patent deviates from the general usage documented, for example, in the attachment by Smith et al. (Bark and ERB Bilinear Transforms, IEEE Transactions on Speech and Audio Processing, Vol. 7, No. 6, November 1999, pp. 697-708; NK5) in that the width of the frequency bands formed in the embodiments differs from NK5 in that it is not aligned with the frequency resolution of the human ear.

65 Whether this means, as the Patent Court believes, that the usual technical usage is also not relevant with regard to all other aspects of the interpretation of feature 1.2 does not need to be decided.

66 As will be shown below, the contested judgment proves to be correct even if it is assumed in favor of the defendant that a division into several frequency bands is absolutely necessary.

67 (3) The Patent Court rightly assumed that feature 1.2 does not necessarily require a conversion from the time domain to the frequency domain, as provided for in claim 2, or even a discrete short-time Fourier transform (short-time discrete Fourier transform, paragraph 32) or a fast Fourier transform (paragraph 36), as mentioned in the description as examples.

68 These procedures are suitable for dividing into multiple frequency bands. However, they are not the only methods that can be used for this purpose. Since feature 1.2 does not contain any specifications in this regard, all methods suitable for this purpose are included.

69 c) The use of a filter that models late reverberation (LR) as provided for in feature 1.3 has the objective of improving the coherence value of the input signals generated in accordance with feature 1.2.

70

aa) According to the description, late reverberation occurs when a listener not only perceives auditory events from different distances, but is also surrounded by diffuse sound, for example in a concert hall where late reverberation reaches their ears from all directions. This auditory experience can be reproduced by using suitable filters (paragraph 54-56; para. 59).

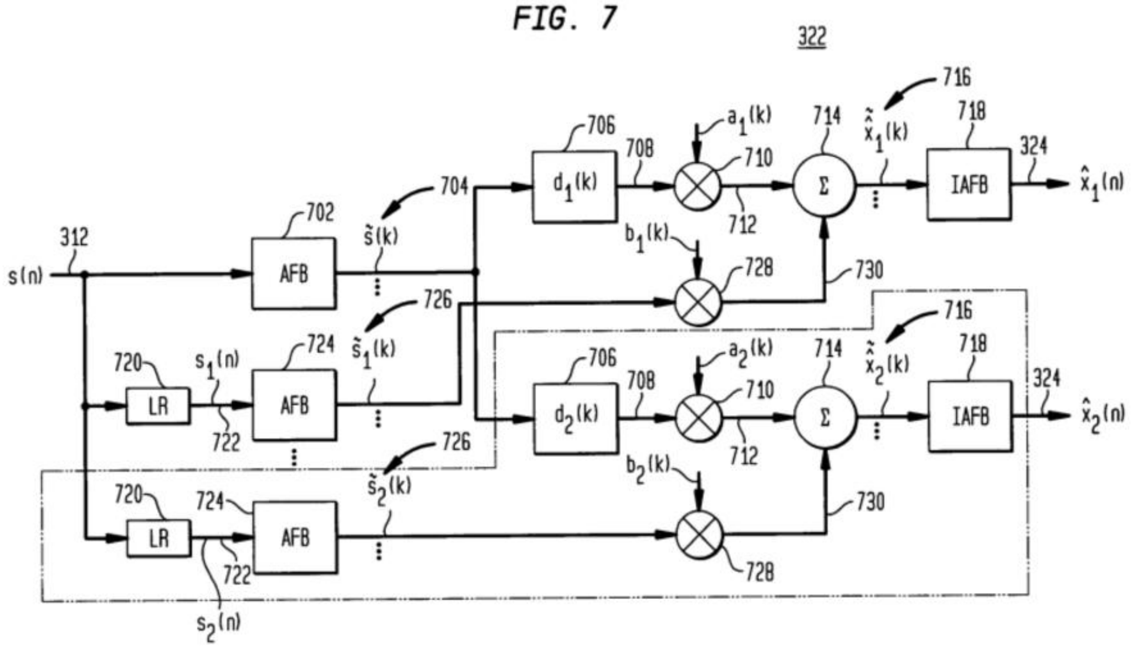
71 bb) Feature 1.3 does not specify which specific type of filter is to be used for this modeling. This means that any filter that can model late reverberation in this sense can be considered.

72 cc) Feature 1.3 also leaves open whether the filter is to be applied to the input signal in the time domain or after its transformation in the frequency domain.

73 This is consistent with the description, which describes both embodiments.

74 (1) In the embodiment shown in Figure 7 below, the filter is applied to a signal in the time domain.

FIG. 7

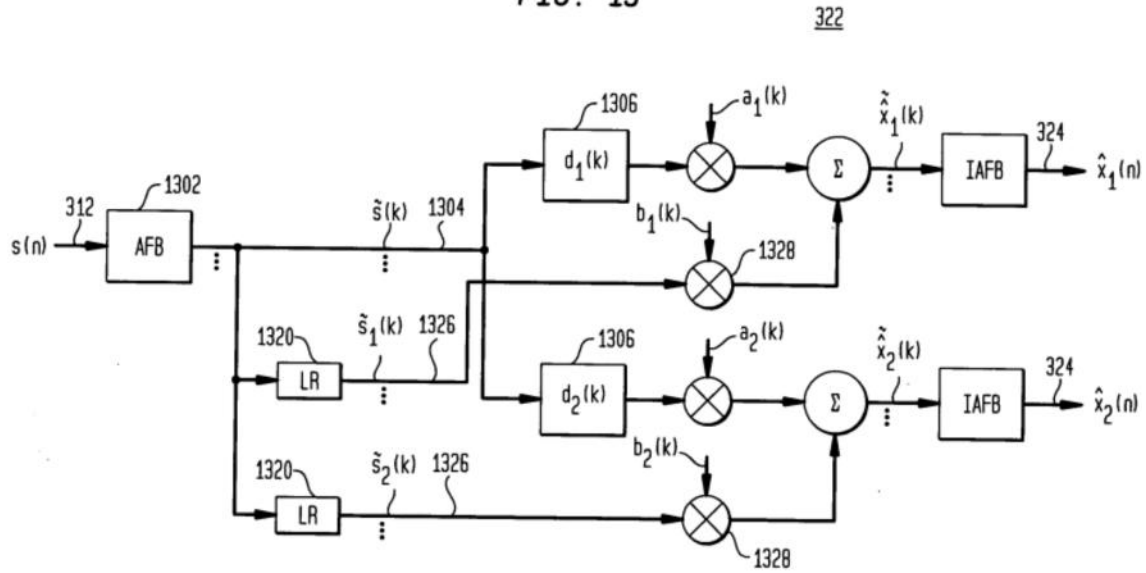


75 Similar to the embodiment shown in Figure 5, the input channel (312) is converted by an auditory filter bank block (702) into two copies of a signal (704) in the frequency domain. Both copies are delayed by a delay block (706) and scaled by a multiplier (710) (paragraph 61).

76 Two further copies of the input channel (312) are each fed to a processor (720) for generating late reverberation and processed there in a suitable manner (paragraph 62). The diffuse channels (722) generated in this way are each fed to an auditory filter bank block (724). The blocks (702, 724) preferably generate sub-bands whose bandwidth is equal to or proportional to the critical bandwidths of the ear (paragraph 64). The LR signals (726) generated by the block (724) are each scaled by a multiplier (728) with factors derived from the additional information processed by the processor (318) (paragraph 65). The resulting LR signal (730) is added by a summing node (714) to the corresponding signal (712) generated by the multiplier (710) (paragraph 66).

77 (2) In the embodiment shown in Figure 13 below, the filter is applied to model the reverberation of a signal in the frequency domain.

FIG. 13


322
s(n)
312
AFB
1302
$\tilde{s}(k)$
1304
LR
1320
$\tilde{s}_1(k)$
1326
LR
1320
$\tilde{s}_2(k)$
1326
$d_1(k)$
1306
$d_2(k)$
1306
$a_1(k)$
$b_1(k)$
1328
$a_2(k)$
$b_2(k)$
1328
Σ
Σ
$\tilde{\hat{x}}_1(k)$
$\tilde{\hat{x}}_2(k)$
IAFB
IAFB
324
324
$\hat{x}_1(n)$
$\hat{x}_2(n)$

78 In this embodiment, the auditory filter bank block (1302) generates four copies of a signal in the frequency domain. Two of these copies are fed to a delay block (1306) and a multiplier, respectively, as shown in Figure 7, while the other two are fed to a processor (1320) for generating late reverberation. The signals generated in this way are combined in the same manner as in Figure 7 (paragraph 91).

79 In this embodiment, different filter lengths can be used for different frequency bands, for example, shorter filters for higher frequencies. This reduces the computational complexity (paragraph 92).

80 dd) Feature 1.3 also does not exclude subjecting the input channel to additional measures prior to filtering. In particular, it is not excluded to combine the input channel with further audio signals and to feed this combined signal to the filtering provided for in feature 1.3.

81 The requirement that at least one input channel be filtered in this manner expressly opens up the possibility of filtering multiple input channels. However, it does not specify that each input channel must be filtered separately.

82 d) Feature 1.4 does not specify the procedure shown in Figures 7 and 13 in every detail.

83 aa) According to feature 1.4, it is necessary, but also sufficient, that each of the LR-filtered diffuse signals generated according to feature 1.3 be combined with a corresponding processed input signal generated according to feature 1.2 in order to generate one of several output channels for the auditory scene.

84 bb) As the Patent Court correctly assumed, this presupposes that the number of diffuse signals corresponds to the number of processed input signals. The embodiments shown in the description as possible alternatives, in which individual output channels are generated without reverberation or individual LR filters are used to generate multiple output channels (paragraph 97), are therefore excluded.

85 The fact that each processed input signal within the meaning of feature 1.2 must be combined with an LR-filtered diffuse signal within the meaning of feature 1.3 already follows from the inlet wording in feature 1.4, according to which the defined requirement applies to each of these signals. This excludes the possibility of feeding individual processed input signals to the output channels without reverberation.

86 The fact that each LR-filtered diffuse signal may only be combined with one processed input signal follows from the combination of features 1.2 and 1.3, according to which two or more processed input signals and corresponding LR-filtered diffuse signals must be generated. It can be inferred from this that the number of processed input signals and the number of LR-filtered diffuse signals must be equal. This requires that the combination of each LR-filtered diffuse signal with a corresponding processed input signal specified in feature 1.4 must be carried out by means of a 1:1 assignment.

87 cc) However, as the plaintiff rightly argues in essence and as the Patent Court has also accepted, features 1.2 to 1.4 do not preclude a processed input signal or an LR-filtered diffuse signal from being composed of several partial signals generated by using different filters.

88 This follows from the fact that features 1.2 and 1.3 do not specify in detail how the filtering is to be carried out. This leaves open the possibility of using

several filters of this type in parallel and composing the processed input signal or the LR-filtered diffuse signal from several partial signals obtained in this way.

89 6. The features of the device protected in claim 8 essentially correspond to features 1.1 to 1.4 of claim 1. Both claims are therefore subject to the same assessment with regard to most aspects.

90 Unlike feature 1.2, however, feature 8.2 does not provide for an auditory filter bank block, but only for unspecified means for generating two or more processed input signals.

91 7. Despite some differences in wording, the device protected by patent claim 9 corresponds in many features to the device according to claim 8.

92 A decisive difference is that features 9.2 and 9.4 mandatorily provide converters for converting from the time domain to the frequency domain and vice versa.

93 II. The Patent Court essentially gave its reasons as follows:

94 The patent in suit rightly claims priority from US patent application 10/815591 (BP6) dated April 1, 2004. The initial applicants transferred all rights from this application to the defendant's legal predecessor.

95 International patent application 99/14983 (NK2) completely anticipates the subject matter of patent claims 1 and 8. The subject matter of claim 9 is suggested by NK2 in conjunction with general technical knowledge or NK9. A skilled person, a physicist, electrical engineer, or sound engineer with a university degree and several years of practical experience in the coding and other processing of audio signals and data to generate various auditory impressions/scenarios using

suitable output means, would know that an audio signal can be converted from the time domain to the frequency domain in order to be processed, and must then be converted back in order to be output at the output channel.

96 Accordingly, the subject matter of claims 1 and 8 formulated in auxiliary request 1, which also necessarily provide for conversion from the time domain to the frequency domain, is also suggested. The same applies to the subject matter of claims 1, 7, and 8 according to auxiliary request 2.

97 The subject matter defended in auxiliary request 2A, which provides for the delaying of each FD input signal with delay values derived from corresponding inter-channel time difference(ICTD) data, and scaling the delayed FD input signals by a corresponding multiplier with scaling factors derived from the corresponding inter-channel level difference (ICLD) data, is new compared to the prior art in the proceedings.

98 NK2 provides no reason to provide data processing steps based on time and intensity difference data of different channels in the frequency domain. NK9 does not deal with the processing of a late echo and therefore gives no reason to include signal components of the late echo in considerations regarding signal processing in the frequency domain.

99 US Patent 5 371 799 (NK3) does describe processing steps within the meaning of features 1.1 to 1.4. However, the citation does not give rise to the derivation of the delay and scaling values specified in auxiliary request 2A, nor to the sequence of delaying and scaling specified therein.

100 The remaining citations are further removed and are therefore unsuitable as a starting point for technical considerations or in a possible

overview to arrive at the subject matter of claim 1 according to auxiliary request 2A in an obvious manner.

101 III. This assessment stands up to review in the appeal proceedings, at least in terms of the result.

102 1 The Patent Court correctly assumed that the dissertation published in September 2004 by Christoff Faller, the co-inventor named in the contested patent (Parametric Coding of Spatial Audio, École Polytechnique Fédérale de Lausanne, Thèse No. 3062, September 2004, NK1) should not be used to assess novelty because the contested patent effectively claims the priority of application BP6, filed on April 1, 2004.

103 a) It is undisputed between the parties that BP6 discloses the same invention as the contested patent.

104 b) The Patent Court correctly concluded that the applicant for the contested patent was entitled to claim priority because the applicants for BP6, who are also named as inventors in the contested patent, had assigned this right to it.

105 aa) After issuing the contested judgment, the Enlarged Board of Appeal of the European Patent Office ruled that there is a rebuttable but strong presumption in favor of entitlement to claim priority (EPO, decision of October 10, 2023 - G 1/22, para. 86, para. 101 et seq. and para. 122 – Prioritätsberechtigung).

106 The Senate concurred with this view (BGH, judgment of November 28, 2023 - X ZR 83/21, GRUR 2024, 374 para. 110 et seq. - Sorafenib-Tosylat; judgment of January 9, 2024 - X ZR 74/21, GRUR 2024, 603 para. 67 et seq. - Happy Bit).

107 Accordingly, in the event of a dispute, it is incumbent upon the plaintiff to rebut the presumption.

108 bb) The plaintiff's submission does not meet these requirements.

109 (1) The declaration (G5) submitted with application BP6, in which the two applicants assigned their rights to the applicant of the contested patent and authorized the latter to apply for a patent for the invention outside the United States, claiming the priority of BP6, does not rebut the presumption; it even constitutes additional evidence of an effective transfer of the priority right.

110 The fact that the signatures of the two applicants in G5 – like the signatures in BP6 – are reproduced on two separate pages containing different fax headers does not preclude the validity of the declarations. What is decisive is not the form in which the declarations were submitted to the United States Patent and Trademark Office, but the form in which they were made. The transmission by fax documented in G5 suggests that the declarations were signed in the original.

111 The fact that the original signatures may have been affixed to separate copies of the transfer declaration does not preclude the validity of the declarations either.

112 (2) The presumption in favor of the applicant for the contested patent is further strengthened in the present case by the fact that both applicants for BP6 have reconfirmed in writing in connection with the present legal dispute that they signed the transfer declaration on the date specified in G5 (G7, G8).

113 (3) Against this background, the plaintiff's submission does not even provide reasons for serious doubts as to the validity of the transfer. It is certainly not sufficient to refute the presumption in favor of the applicant for the contested patent.

114 (4) In this situation, the order requested by the plaintiff for the defendant or the applicant for the disputed patent to submit the original declaration cannot be considered.

115 The submission of the original would - assuming the authenticity of the signatures - provide full proof that the declarations contained therein had been made, in accordance with Section 416 of the German Code of Civil Procedure (ZPO). This proof is not required in the circumstances of the present case.

116 If the document were not submitted despite the order, this could provide reasons for doubts as to whether the declarations had been made. However, in view of the circumstances outlined above, this would not be sufficient to refute the presumption.

117 cc) Contrary to the plaintiff's view, the defendant has no further burden of proof.

118 The extent to which the applicant for the contested patent has a secondary burden of proof in the constellation in question can be left open. In the case in dispute, the defendant has explained the process by which the right to claim priority was transferred. It has also argued that the applicants for BP6 confirm this statement. The defendant is not required to provide any further explanations.

119 2. The Patent Court rightly assumed that NK2 completely anticipates the subject matter of claim 8.

120 a) NK2 describes sound reproduction techniques with improved spatial effects (p. 1, lines 3 et seq.; p. 2, line 5).

121 aa) The description of NK2 explains that, for a more pleasant listening experience via a pair of headphones, it is desirable to reproduce the atmosphere of the original recording spatially, preferably in such a way that the impression is created that the sound is coming from outside the listener's head. With

conventional techniques for generating three-dimensional sound, this impression is lost when using standard headphones (p. 1, lines 7-21).

122 In the prior art, it is known to fold audio signals with suitable head-related transfer functions (HRTF). However, this often requires excessive computing resources (p. 1, lines 27-29).

123 bb) To improve this, NK2 proposes converting an audio signal intended for reproduction via loudspeakers into a specific signal for each ear by means of different filtering.

124 In the simplest embodiment, the signal is passed through two filters, each of which is assigned to one of the two ears (p. 5, lines 30-32).

125 In more complex embodiments, the signals intended for individual loudspeakers are filtered individually in the manner described and then combined to create one signal for each ear. The filters represent the head-related transfer functions (HRTF) (p. 5, lines 38 to p. 6, lines 3; Figure 2).

126 cc) An embodiment in which five channels, as known from the Dolby AC-3 standard, are processed in this manner is shown schematically in Figure 4 below.

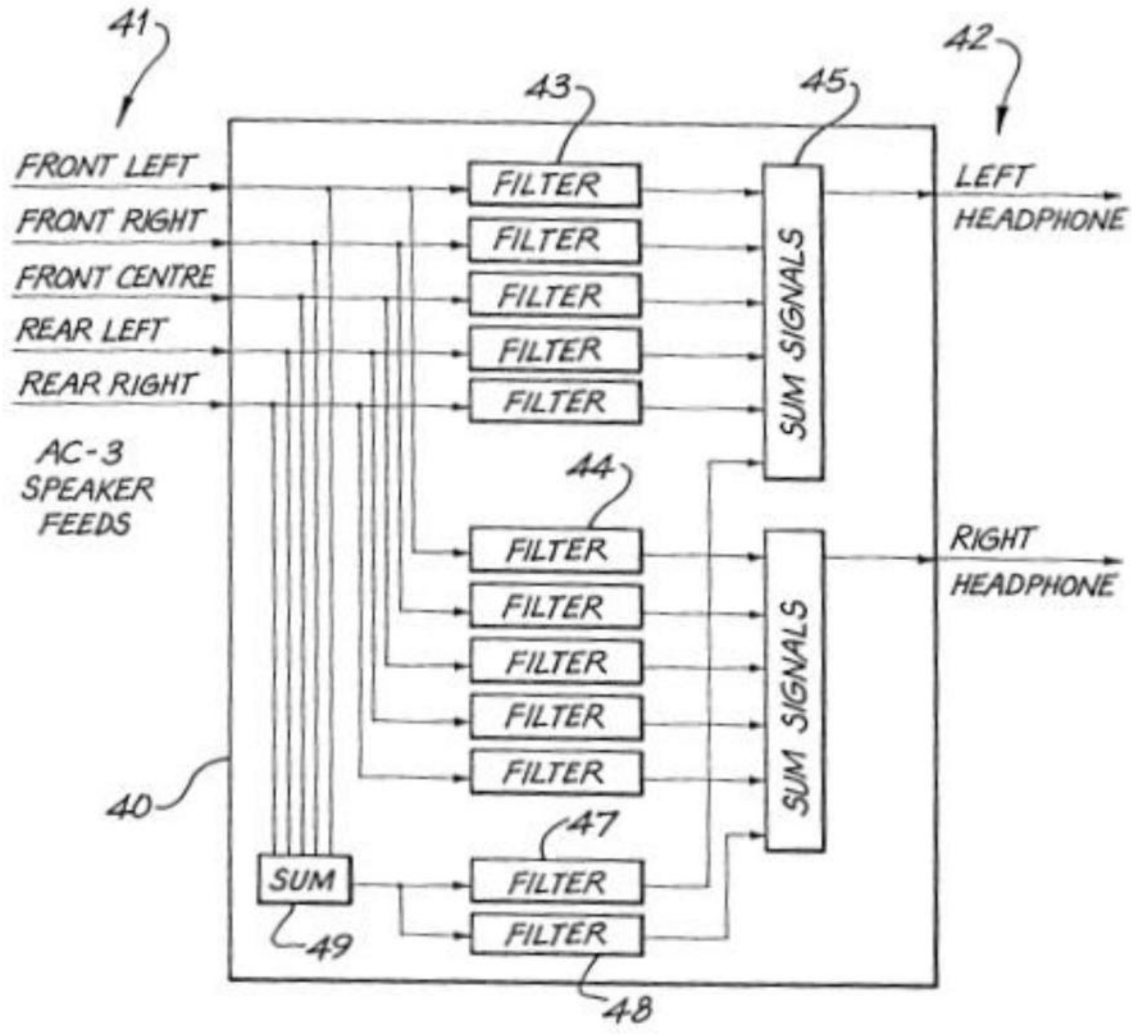


FIG. 4

127 Each of the signals (41) intended for the five loudspeakers is passed through a filter (43, 44) for the right and left ears, respectively. The filtered signals are summed separately for each ear (45) in order to generate a headphone signal for each output channel (p. 6, lines 23-27).

128 In order to achieve high quality and take into account the spatial geometry of the listening environment, long filters must be used. However, depending on the application environment, the processing requirements may be too high (p. 6, lines 30-34).

129 To reduce the required computing power, ten short filters (43, 44) and only two full-length filters (47, 48) can be used. The longer filters (47, 48) can be a binaural simulation of the tail of an average room response, and the shorter filters can be the early part of the response (p. 6, lines 35-39).

130 These signals are combined (45) with the headphone signals processed and summed for each output channel via the filters (43, 44) and fed to the output channel.

131 dd) Finite impulse response (FIR) filters can be used to generate reverberation. In this case, it may be advantageous to feed the output of these filters back into the virtual speaker input (p. 9, lines 22-26).

132 However, the use of very long FIR filters requires large storage capacities. Infinite impulse response (IIR) filters require less memory and therefore usually less computing power. However, their use can impair the spatial impression (p. 11, lines 19-23).

133 To account for this, higher-quality processing can be provided for the early part of the simulated acoustic response—for example, for the direct sound and some early reflections (p. 11, lines 24-28).

134 Examples of this are shown schematically in Figures 16 and 17 below.

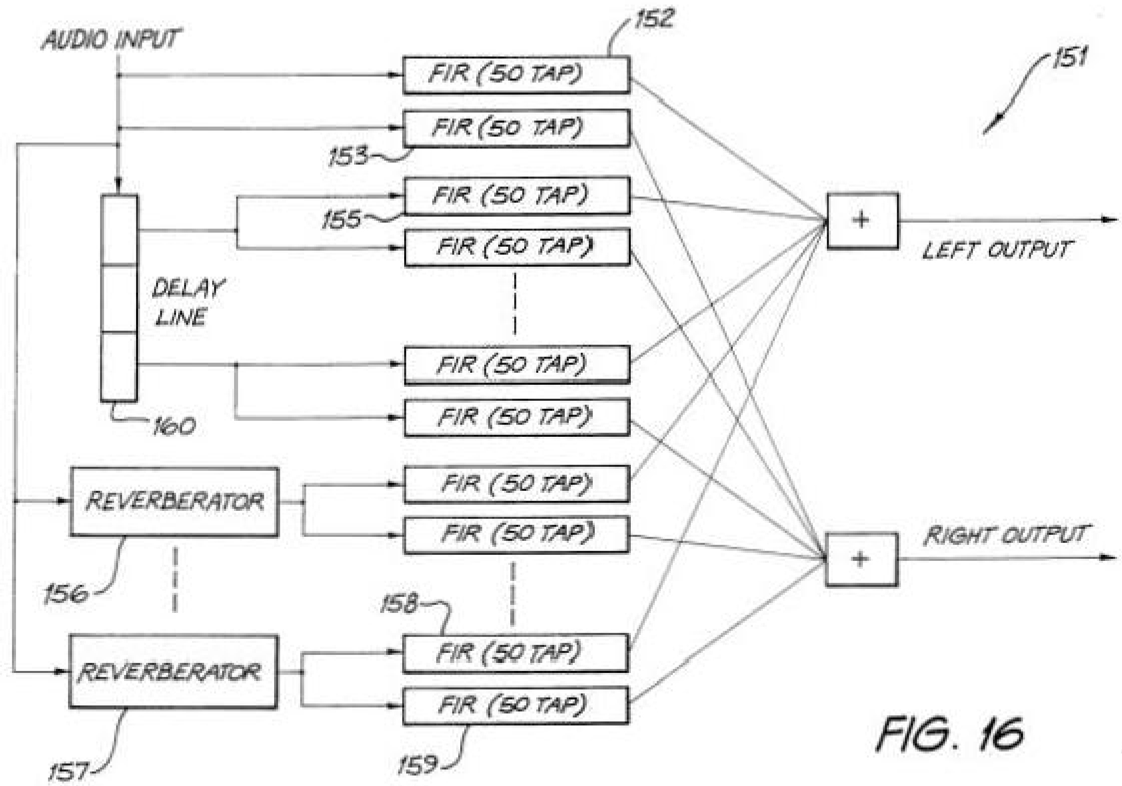

AUDIO INPUT
152
151
FIR (50 TAP)
153
155
DELAY LINE
160
LEFT OUTPUT
RIGHT OUTPUT
REVERBERATOR
156
157
158
159
+

FIG. 16

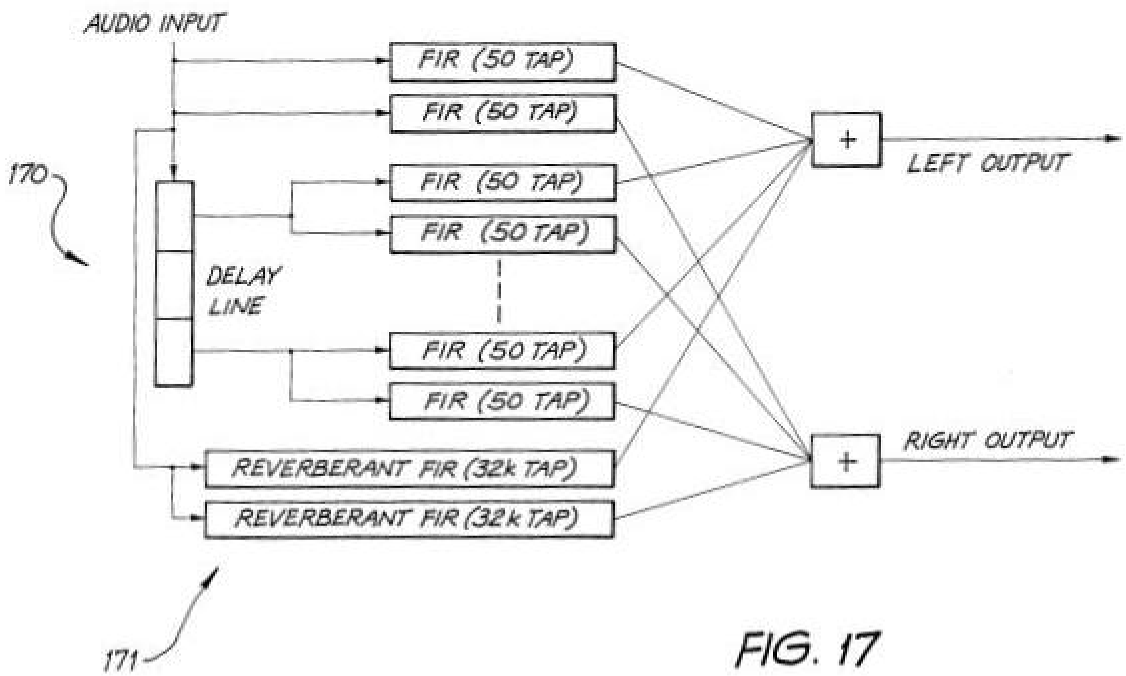

AUDIO INPUT
170
FIR (50 TAP)
DELAY LINE
LEFT OUTPUT
RIGHT OUTPUT
REVERBERANT FIR (32k TAP)
171
+

FIG. 17

135 In both examples, a direct signal and a signal with a short delay are branched off from the input channel and routed through several pairs of FIR filters, each with 50 taps.

136 To generate late reverberation (late reverberant part), two reverberation generators (157) in Figure 16 are also each coupled to an FIR filter with 50 taps (p. 11, lines 29-39).

137 In Figure 17, a pair of long reverberant FIR filters (171) is provided instead. This allows a much more accurate spatial impression to be generated than by the use of recursive reverberant structures shown in Figure 16 (p. 12, lines 7-12).

138 b) This discloses all the features of claim 8.

139 aa) As the defendant does not dispute, NK2 discloses a device for processing an audio signal in order to synthesize an auditory scene, and thus feature 8.1.

140 bb) NK2 also discloses the generation of at least two processed input signals within the meaning of feature 8.2.

141 In this context, it is irrelevant how the filters used in NK2 are designed in detail. As explained above, feature 8.2- unlike feature 1.2 - does not specify the means for generating the processed output signals.

142 cc) NK2 also discloses the generation of LR-filtered diffuse signals within the meaning of feature 8.3.

143 (1) As the defendant does not dispute, both Figure 4 (with filters (47, 48)) and Figure 17 (with reverberation FIR filters (171)) show filters that model late reverberation and generate a total of two LR-filtered diffuse signals.

144 In the initial example shown in Figure 17, two LR-filtered diffuse signals are created from one input channel using separate filters. This is sufficient to implement feature 8.3.

145 (2) Contrary to the defendant's opinion, feature 8.3 is also realized in the embodiment shown in Figure 4.

146 This is not contradicted by the fact that a separate pair of processed input signals is generated for each of the five input channels, but only one common pair of an LR-filtered diffuse signal.

147 As already explained above, features 1.2 and 1.3 do not preclude combining an input channel with other audio signals prior to filtering. Such a combination also exists when several input channels are combined prior to generating the late reverberation.

148 The same applies to features 8.2 and 8.3.

149 dd) Feature 8.4 is also disclosed in NK2 – both in Figure 4 and in Figure 17.

150 (1) With regard to the embodiment shown in Figure 4, this results from the fact that two processed input signals and two LR-filtered diffuse signals are generated for each input channel and these signals are combined with each other in a 1:1 relationship with respect to the respective input channel.

151 The fact that the two LR-filtered diffuse signals are obtained from a combination of the input channels and are therefore identical for each input channel does not preclude this. As already explained above, the combination of input channels prior to filtering is possible according to both feature 1.2 and feature 1.3. The 1:1 assignment required by feature 1.4 is already given in such constellations

if a processed input signal is combined with a corresponding LR-filtered diffuse signal for each input channel.

152 The same applies to feature 8.4.

153 (2) In the embodiment shown in Figure 17, feature 8.4 is realized because each of the two LR-filtered diffuse signals from the filters (171) is combined with a processed input signal from the other filter groups assigned to it.

154 The fact that the two processed input signals are composed of several partial signals is harmless because features 1.2 and 8.2 do not preclude such a design for the reasons already explained.

155 3. The Patent Court correctly assumed that NK3 also completely anticipates the subject matter of claim 8.

156 a) NK3 deals with the processing of audio signals intended for loudspeaker systems for playback via headphones, in which the sounds appear to the listener as if they were coming from a source located outside the head at a point in the space around the listener (col. 2, lines 4 et seq.).

157 aa) The description of NK3 explains that an audio signal is basically divided into three parts. These are shown schematically in Figure 1 below.

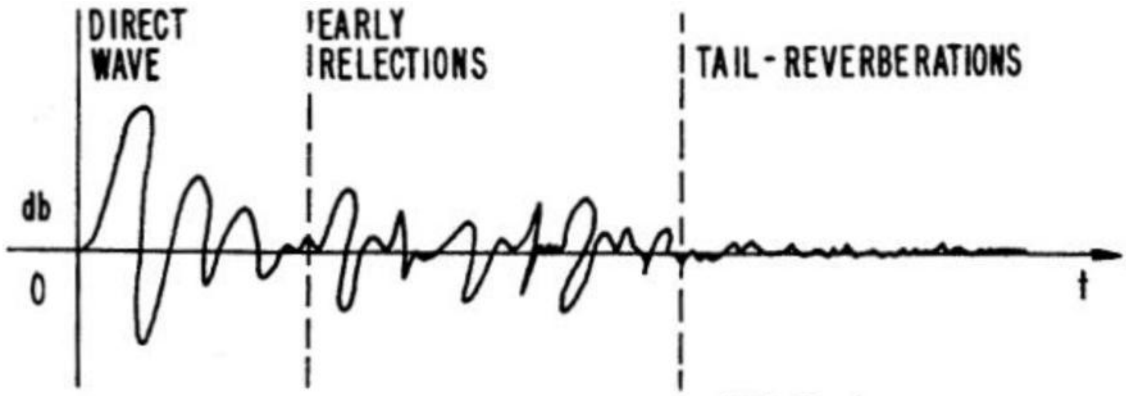


FIG. 1

158 The first component is the direct wave component, which represents the sound that arrives directly at the listener's ear. The second component consists of a series of early reflection portions, which represent the original signal reflected from the walls, floor, and ceiling of the room in which the listener is located. The third component represents the tail or reverberation, which consists of multiple reflections of the sound wave after it has been reflected several times by the walls, floor, and ceiling, so that the directional information is completely incoherent (col. 3, lines 27 et seq.).

159 bb) An approach to developing a transfer function that represents a sound wave of the type shown in Figure 1 is illustrated in Figure 2 below.

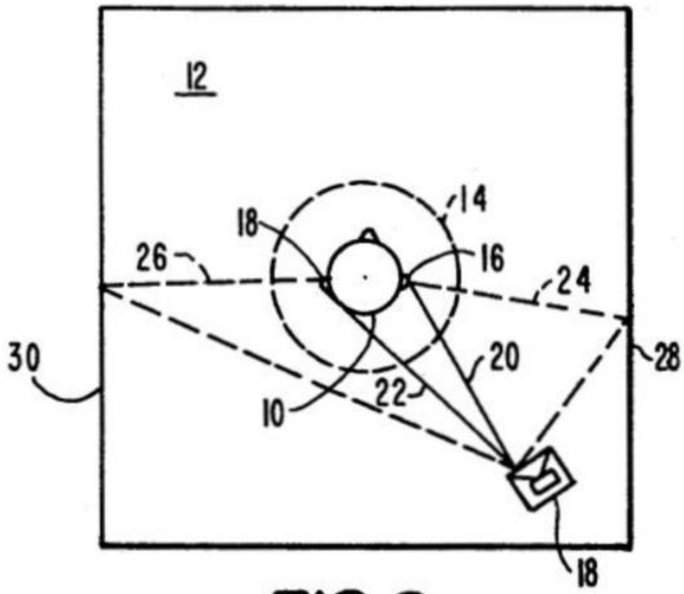


FIG. 2

160 A loudspeaker (18) can be positioned so that the sound reaches the ears both via direct paths (20, 22) and via reflection paths (24, 26). By moving the loudspeaker to different locations in the listener's environment and determining the waveforms with microphones (16, 18) at the right and left ears, a library of sound positions can be created. On this basis, according to the invention disclosed in NK3, each input signal can be processed to simulate a source position that corresponds to one of the stored patterns (col. 3, line 46 to col. 4, line 8).

161 cc) However, such a large filter is not practical for a marketable product. NK3 therefore proposes a more economical system, which is schematically represented in Figure 3 below (col. 4, lines 8-11).

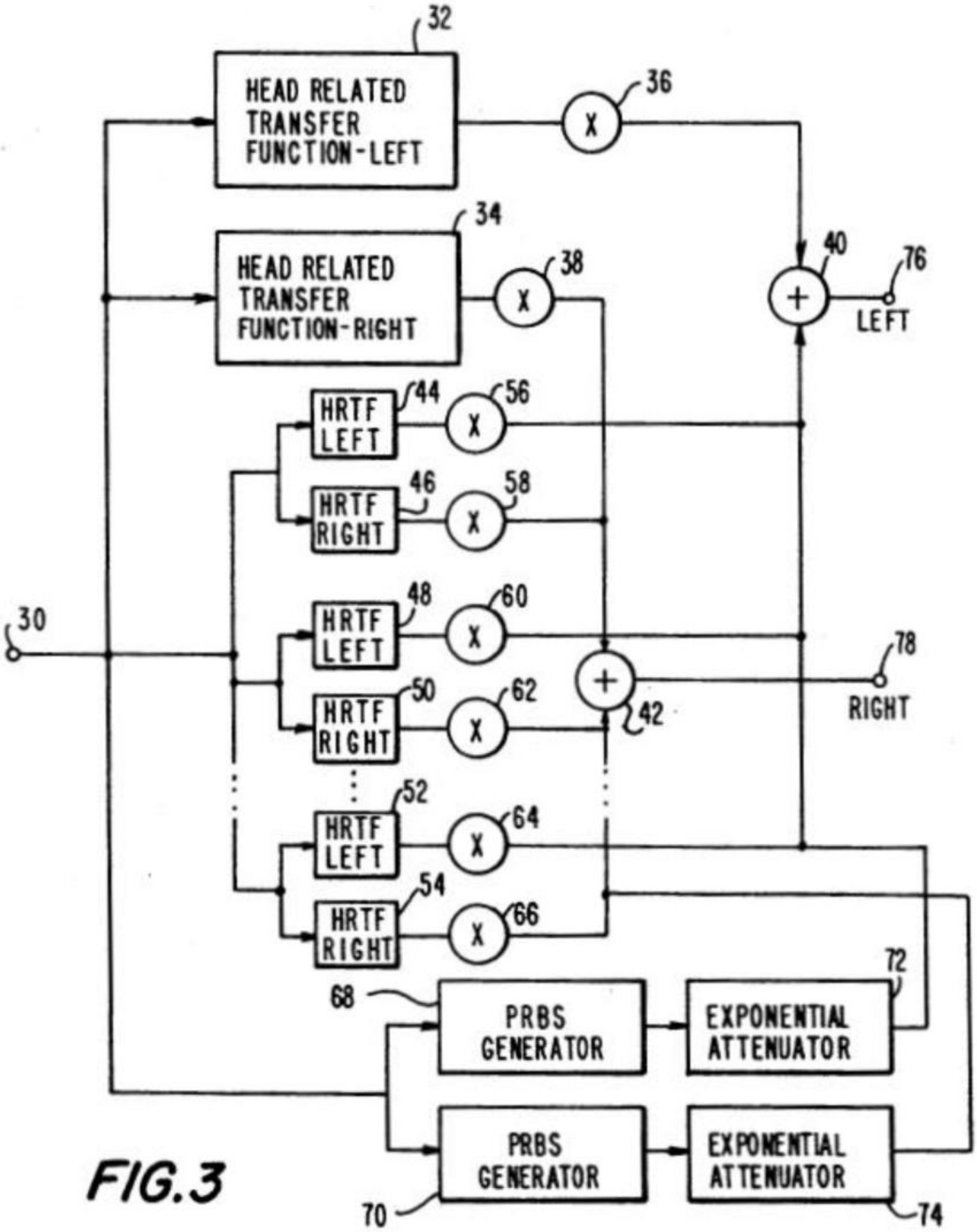


FIG.3

162 The audio signal (30) to be processed is fed to two devices (32, 34) that perform head-related transfer functions (HRTF) for the left and right sides of the head, respectively. These devices are finite impulse response (FIR) digital filters.

They provide transfer functions as can be derived using the system shown in Figure 2. Alternatively, frequency-dependent phase and amplitude filters can be used. It has been found that a single HRTF filter can achieve all angular positions (azimuths) over a range of 180° by using the head-related transfer function for a location directly in front of the listener and then adjusting the amplitude and delay to correspond to the indirect sides of the transfer function (col. 4, lines 12-30).

163 The output of the two filters (32, 34) is passed through scalars (36, 38). These add a weighting factor that provides information about the distance between the headphones and the apparent sound source. The scaled direct wave signals are fed to adders (40, 42) (col. 4, lines 37-44).

164 The audio signal (30) is further passed through several filter pairs (44/46, 48/50, 52/54). These primary or secondary reflection filters can be significantly shorter than the filters (32, 34) used for the direct wave component (col. 4, lines 44-52). The filtered signals are also passed through scalars (56/58, 60/62, 64/66) and then fed into the adders (40, 42) (col. 4, lines 62 to col. 5, lines 9).

165 For the reverberation component, the audio signal (30) is fed to a pair of pseudo-random binary sequence generators (68, 70). Their outputs are routed through scalar or exponential attenuators (72, 74) and fed to the left and right channel signals in the adders (40, 42) (col. 5, lines 10-29).

166 In this way, three sound components are filtered or simulated and then combined in the adders (40, 42) so that a right or left headphone channel is available at the outputs (76, 78) of the adders (col. 5, lines 30-36).

167 dd) The operation of the processors for adaptation to angle ranges (azimuth ranges) is schematically illustrated in Figure 7 below, among other places.

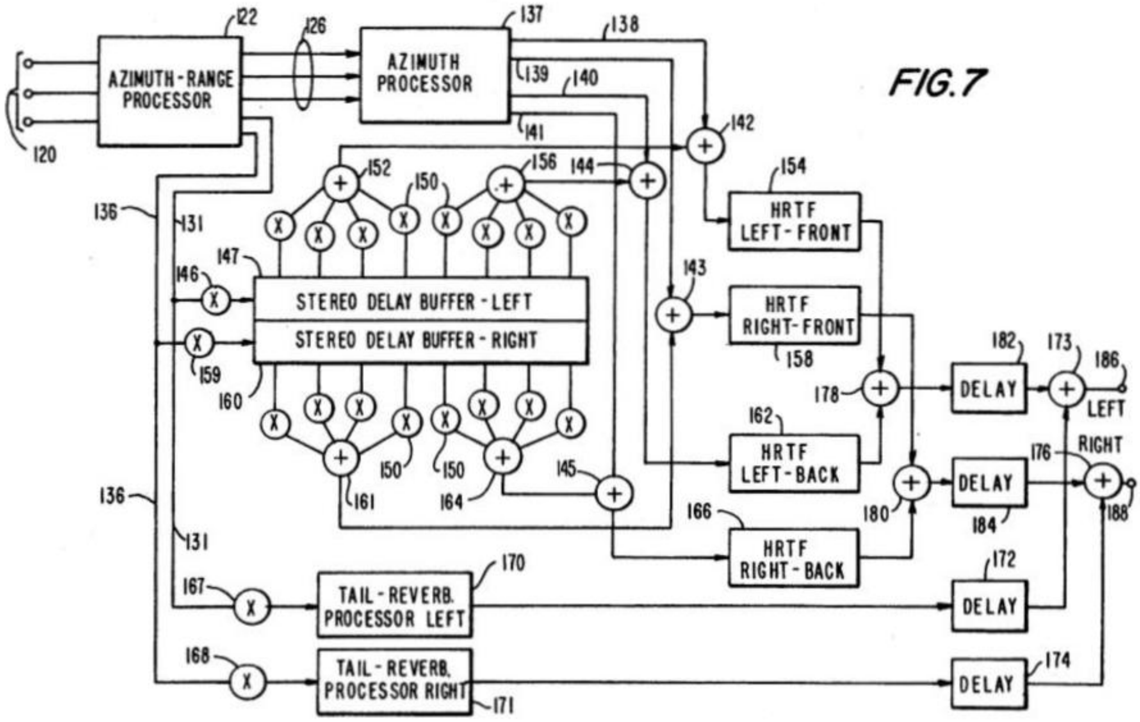


FIG.7

168 The input samples (120) are fed into an azimuth range processor (122). There, the inputs (120) are scaled and summed to form two reverberation channels (131, 136) (col. 6, lines 43-64).

169 The input samples (120), which represent the direct wave component, are scaled separately and passed on to an azimuth processor (137) via lines (126). This applies values from a delay and amplitude table and generates four signals for front right/left and rear right/left (col. 6, lines 65 to col. 7, lines 14).

170 The signals representing the first reflections are fed to stereo delay buffers (147, 160) for the left and right channels via scalars (146, 159).

The appropriately scaled first reflections are summed at four points (152, 156, 161, 164) and fed via adders (142, 143, 144, 145) to the four signals of the direct wave component (col. 7, lines 14-51). The two signals for the left and right sides are combined by adders (188, 180) so that one signal remains for each side.

171 The signals for the reverberation (131, 136) are each fed to a scaler (167, 168) and a flag filter or reverberation processor (170, 171). They are then fed via a delay unit (172, 174) to adders (173, 176), which combine them with the other signal for the left or right side (col. 7, line 52 to col. 8, line 2).

172 b) This also discloses the features of claim 8.

173 aa) As the defendant does not dispute, NK3 discloses a device within the meaning of feature 8.1 and filters for modeling late reverberation within the meaning of feature 8.3.

174 bb) Feature 8.2 is also disclosed.

175 As already explained above, it is sufficient for this purpose that at least two processed input signals are generated. This requirement is met by the system proposed in NK3.

176 cc) Contrary to the defendant's opinion, NK3 also discloses feature 8.4.

177 As already explained in connection with NK2, this is not contradicted by the fact that in both Figure 3 and Figure 7, the LR-filtered diffuse signals are generated from a sum signal, while the processed input signals are generated separately for each channel and only combined in a second step.

178 4. It can be left open whether, as assumed by the Patent Court, the subject matter of claim 1 is also disclosed in NK2. In any case, this subject matter was obvious based on NK2 in conjunction with general technical knowledge.

179 a) Features 1.1, 1.3, and 1.4 are disclosed in NK2 for the same reasons as features 8.1, 8.3, and 8.4.

180 b) If, in favor of the defendant, it is assumed that feature 1.2 requires a division into frequency bands, this feature is not disclosed in NK2.

181 As the defendant rightly argues, and as the Patent Court also accepted in connection with the granted version of claim 9 and auxiliary requests 1 and 2, NK2 does not contain any direct and unambiguous information on the nature of the filters used.

182 NK2 does indeed use the term "filter banks" in connection with Figure 2 at one point (p. 6, line 10), as the plaintiff correctly points out. However, it cannot be inferred with the necessary clarity that the term is used in the technical sense as defined, for example, in Mertins' textbook (Signal Theory, 1st edition 1996, G10'), or whether it merely refers to a parallel arrangement of filters for generating four different signals, as shown in Figure 2.

183 c) However, as the Patent Court correctly pointed out in connection with claim 9, it was obvious based on general technical knowledge to design the filters used in NK2 to generate the processed input signals in such a way that they convert the signal for processing from the time domain to the frequency domain and, after processing, perform a conversion in the opposite direction.

184 aa) As the plaintiff rightly argues, NK2 was a logical starting point for further consideration simply because it does not explain the design of the proposed filters in detail.

185 In this context, it is irrelevant whether there was reason to use the system disclosed in NK2 in BCC decoders, such as those known from NK9. Even a skilled person who wanted to use such a system only for the purpose described in NK2—the conversion of a signal intended for loudspeakers into a signal for headphones—was faced with the question of how to design the filters proposed in NK2 so that the objective specified there could be achieved. This gave rise to a search for suitable filtering options in the prior art.

186 bb) According to the findings of the Patent Court, it was common knowledge that it may be expedient to convert a signal from the time domain to the frequency domain for the purpose of filtering.

187 The defendant does not provide any concrete evidence to cast doubt on the completeness and accuracy of this finding.

188 As the plaintiff rightly points out, the findings made are confirmed by the explanations in the textbook by Jeruchim et al. (Simulation of Communication Systems, Second Edition, 2002, NK11). Although this deals with a more abstract issue. The inlet remarks, according to which it is of crucial importance from the point of view of simulation techniques whether the impulse response has a finite (FIR) or infinite duration (IIR), and that FIR filtering usually takes place in the frequency domain because the Fast Fourier Transform is available here (p. 9, paragraph 2), are of interest from the perspective of NK2 precisely because they do not refer to specific applications and because NK2 describes the use of FIR filters as preferable.

189 This gave rise, as the Patent Court rightly assumed in its conclusion, to sufficient grounds for considering frequency domain filtering using a fast Fourier transform for a system based on the NK2 model. In such a design, feature 1.2 is also realized in the interpretation postulated by the defendant.

190 cc) Contrary to the defendant's opinion, a different assessment is not required even if NK2 - as the defendant asserts - describes exclusively filtering in the time domain.

191 The description of NK2 indicates that the desired functions can be achieved with different types of filters. Differences between FIR and IIR filters are described, and FIR filters are described as preferable. However, even in this respect, no design is excluded.

192 The description of NK2 also does not provide any indication that only filtering in the time domain is possible. According to the defendant's submission, filtering in the examples of implementation takes place in the time domain. However, NK2 does not explicitly address this aspect and does not contain any other indications that this is absolutely necessary.

193 Against this background, the indication in NK2 that FIR filtering may be advantageous gave rise to a search for designs of this filter technology that are common in the prior art. Based on this, the above-mentioned technical knowledge suggested considering the conversion to the frequency domain that is frequently performed in this field, in particular in the form of a fast Fourier transform. No specific aspects that would make this appear impracticable or difficult have been identified or are otherwise apparent.

194 dd) Since the signal generated in NK2 is to be output to headphones, it was also obvious, given the background described above, to perform converting from the frequency domain to the time domain after processing, as provided for in feature 9.4.

195 5 For the same reasons, the subject matter of claims 1 and 9 was also obvious based on NK3.

196 NK3 also describes FIR filters as preferable, but does not deal in detail with the design of such filters and the question of whether filtering should take place in the time or frequency domain. Based on this citation, it was therefore also obvious to consider filtering in the frequency domain and conversion using a fast Fourier transform.

197 Contrary to the defendant's view, a different assessment does not result from the fact that NK3 describes an embodiment in which the processing is carried out in the time domain (col. 4, lines 53-59). Like NK2, NK3 contains no indication that such a design is absolutely necessary. As the plaintiff rightly argues, the description of Figure 3 even expressly points out that the HRTF filters provided therein can be replaced by frequency-dependent phase and amplitude filters (col. 4, lines 19-21).

198 6. The Patent Court rightly concluded that the subject matter defended by auxiliary request 1 was suggested by the prior art.

199 a) According to auxiliary request 1, features 1.2 and 8.2 are to be supplemented as follows:

200

1.2'	processing (702) at least one input channel (312), using an auditory filter bank block (702), performing a time-frequency transform, to generate two or more processed input signals (704);	Verarbeiten (702) von mindestens einem Eingangskanal (312), unter Verwendung eines Auditiv-Filterbank-Blocks (702), unter Durchführung einer Transformation vom Zeit- in den Frequenzbereich, um zwei oder mehr verarbeitete Eingangssignale (704) zu erzeugen;
8.2'	means (702) for processing at least one input channel (312), using an auditory filter bank block (702) performing a time-frequency transform, to generate two or more processed input signals (704);	Mittel (702) zum Verarbeiten von mindestens einem Eingangskanal (312), unter Verwendung eines Auditiv-Filterbank-Blocks (702), unter Durchführung einer Transformation vom Zeit- in den Frequenzbereich, um zwei oder mehr verarbeitete Eingangssignale (704) zu erzeugen;

201 Claim 9 shall remain unchanged.

202 b) The use of an auditory filter bank block, which is also provided for in feature 8.2' according to this version, was obvious for the reasons already stated in relation to the granted version of patent claim 1.

203 c) The same applies to the conversion from the time domain to the frequency domain provided for in features 1.2' and 8.2'.

204 7. The Patent Court also rightly concluded that the subject matter defended in auxiliary request 2 was obvious.

205 a) According to auxiliary request 2, claim 1 in the version of auxiliary request 1 is to be supplemented by the following features:

206

1.5	the method further comprising: converting (702) the at least one input channel (312) from a time domain into a frequency domain to generate a plurality of frequency-domain (FD) input signals (704); and	Das Verfahren umfasst ferner: Umwandeln des mindestens einen Eingangskanals aus einem Zeit- in einen Frequenzbereich, um eine Mehrzahl von Eingangssignalen im Frequenzbereich (FD) zu erzeugen.
1.6	wherein processing (702) the at least one input channel (312) comprises: delaying (706) and scaling (710) the FD input signals to generate a plurality of scaled, delayed FD signals (712) as processed input signals.	Das Verarbeiten des mindestens einen Eingangskanals umfasst das Verzögern und Skalieren des FD-Eingangssignals, um eine Mehrzahl von skalierten, verzögerten FD-Signalen als verarbeitete Eingangssignale zu erzeugen.

207 Claim 8, now claim 7, is to be supplemented by corresponding features 7.5 and 7.6. Claim 9, now claim 8, is to remain unchanged.

208 b) Both additional features relate to the generation of the processed input signals within the meaning of feature 1.2.

209 aa) Feature 1.5 clarifies that the processed input signals must be generated in the frequency domain.

210 bb) Feature 1.6 additionally requires that the processing includes delaying and scaling the input signal (already converted to the frequency domain).

211 This requirement applies to the processing of each input signal, as also described in the description of the contested patent (paragraph 37, paragraph 60).

212 (1) Scaling within the meaning of feature 1.6 is a change in the level of a single signal. As the defendant correctly argues, this does not include the summation of two input signals, even if this leads to an increase in level.

213 (2) The order in which the signals are delayed and scaled is not specified in feature 1.6. The relevant statements in the description are not reflected in the claim.

214 The wording "delaying and scaling" does not in itself indicate whether the steps are to be performed in this order. The function assigned to these two processing steps according to the contested patent also does not provide any indication that a specific order must be followed.

215 (3) Feature 1.6 also does not specify the manner or means by which the signals are to be delayed and scaled.

216 c) Feature 1.5 is suggested for the reasons given in relation to granted claims 1 and 9.

217 d) Whether feature 1.6 is disclosed in NK2, as the Patent Court has stated, does not require a final decision. In any case, this design was suggested on the basis of NK3.

218 aa) As stated, NK3 discloses features 1.1, 1.3, and 1.4.

219 bb) Feature 1.2 was obvious based on NK3 for the reasons outlined above. This also suggested feature 1.5.

220 cc) NK3 also discloses in Figure 7 the scaling and delaying of all processed input signals, as provided for in feature 1.6.

221 As already explained above, in the embodiment shown in Figure 7, not only are the signals for early and late reverberation delayed and scaled, but also the signals representing the direct component (126).

222 Again, NK3 does not indicate that these processing steps are performed in the frequency domain. However, this design was obvious for the same reasons as the design according to features 1.2 and 1.5.

223 e) The same applies to the version of claim 7 defended in auxiliary request 2.

224 8. The Patent Court rightly ruled that the subject matter defended in auxiliary request 2A is patentable.

225 a) According to auxiliary request 2A, claims 1, 7, and 8 in the version of auxiliary request 2 are to be supplemented by the following features:

226

1.6.1	wherein each FD input signal is delayed at a corresponding delay block based on delay values derived from corresponding inter-channel time difference data, and	wobei jedes FD-Eingangssignal an einem korrespondierenden Verzögerungsblock auf der Grundlage von Verzögerungswerten verzögert wird, die aus korrespondierenden Daten zur Zeitdifferenz zwischen Kanälen abgeleitet sind;
1.6.2	wherein each delayed FD input signal is scaled by a corresponding multiplier based on scale factors derived from corresponding inter-channel level difference data.	wobei jedes verzögerte FD-Eingangssignal skaliert wird mit einen korrespondierenden Multiplikator, der auf Skalierfaktoren beruht, die aus korrespondierenden Daten zur Pegeldifferenz zwischen Kanälen abgeleitet sind.

227 b) Unlike feature 1.6, this specifies a fixed sequence between the processing steps "delay" and "scale".

228 This sequence results from the fact that feature 1.6.2 specifies scaling of the delayed input signals. This specification requires that scaling takes place after a delay. Additional scaling before the delay is not excluded.

229 c) Features 1.6.1 and 1.6.2 also specify parameters on the basis of which the delay and scaling are to be performed, namely data on the time and level difference between the channels.

230 This presupposes that the input channel from which the processed input signals are generated in accordance with feature 1.2 already contains the information on the time and level difference between the channels.

231 As citations NK2 and NK3 show, such information can also be generated using other parameters, for example by measuring the sound signals arriving at the ears of a fictitious listener.

232 However, the requirement that this must be data makes it sufficiently clear that the information must already be contained in the input channel, as is the case with the additional information belonging to the BCC signal used in the embodiments of the contested patent. On the other hand, it is not sufficient for these values to be calculated on the basis of the input signal or other parameters.

233 However, this does not preclude the consideration of additional parameters during delay and scaling in order to further improve the signal.

234 d) The Patent Court rightly assumed that the subject matter defended in auxiliary request 2A was not obvious based on NK3.

235 aa) Contrary to the plaintiff's opinion, NK3 does not disclose features 1.6.1 and 1.6.2.

236 In NK3, the input signals are also delayed and scaled on the basis of predetermined values. However, these values are not contained in the input signal to be processed but, as the plaintiff correctly argues, have been developed

experimentally in advance. For the reasons outlined above, they therefore do not constitute data within the meaning of features 1.6.1 and 1.6.2.

237 bb) The Patent Court rightly ruled that NK3 did not provide any suggestion to use input signals based on the model of NK9.

238 As the description of the contested patent also explains, NK9 deals with the synthesis of auditory scenes from coded audio data containing BCC parameters, in particular information on level and time differences (interaural level difference, ILD; interaural time delay, ILT). As the contested patent shows, such input channels are also suitable for a method as disclosed in NK3. However, NK3 did not provide any suggestion for such a combination.

239 NK3 does not impose any specific requirements on the input signal. However, the focus is on converting signals for two or more loudspeakers into two signals for headphones. Unlike BCC signals based on the NK9 model, such signals do not require synthesis based on data on level and time differences. In view of this, a suggestion would have been needed to apply the method proposed in NK3 to BCC signals as well.

240 e) Based on NK9, no further suggestion was made.

241 However, since NK9 describes a special possibility for encoding and decoding multi-channel audio signals, there may have been reason to optimize such an audio signal for playback via headphones, as proposed in NK3.

242 However, the method described in NK3 for this purpose consists of first decoding the BCC signal in the manner described in NK9 and then converting the signals thus generated for two or more loudspeakers in the manner described in

NK3. However, none of the citations provide any suggestion for a combination in which the generation of LR-filtered diffuse signals described in NK3 as part of the conversion process is already integrated into the BCC decoding.

243 f) No further suggestions arose from a combination of NK2 and NK9.

244 In NK2, as in NK3, the focus is on converting a signal intended for multiple loudspeakers into two signals for headphones. This citation also did not provide any suggestion for integrating the generation of LR-filtered diffuse signals into the BCC decoding.

245 g) Contrary to the plaintiff's view, US patent specification 5 696 831 (NK12) does not anticipate the subject matter defended in auxiliary request 2A.

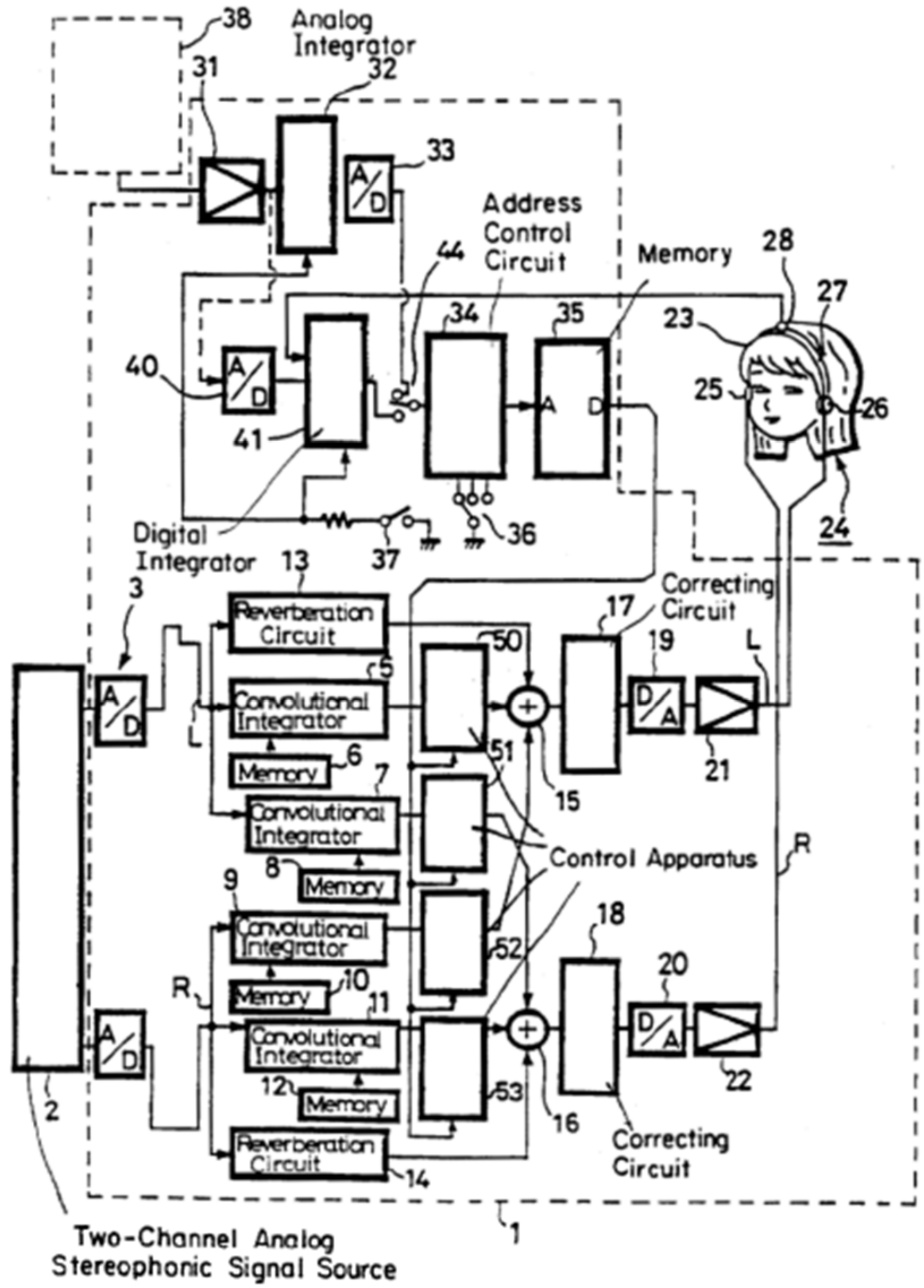
246 aa) NK12 deals with a device for reproducing an audio signal corresponding to an image via headphones.

247 (1) The description of NK12 explains that when using headphones, a phenomenon known as lateralization occurs, whereby the listener perceives the reproduced sound image inside their head even when it originates from a stereophonic signal source (col. 1, lines 11-19).

248 NK12 aims to adjust the sound image so that it matches the image (col. 6, lines 54-58). At the same time, changes in the orientation of the listener's head are to be detected and taken into account by correcting the audio signals (col. 6, lines 64 et seq.).

249 (2) An example of a device designed to achieve these objectives is shown schematically in Figure 2 below.

FIGURE 2


38
Analog Integrator
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33
Address Control Circuit
Memory
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34
35
23
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A/D
A
D
25
26
41
36
37
Digital Integrator
24
13
3
Correcting Circuit
17
19
Reverberation Circuit
50
L
5
Convolutional Integrator
D/A
Memory
6
51
21
7
15
Convolutional Integrator
Control Apparatus
R
8
Memory
9
18
Convolutional Integrator
52
20
R
Memory
10
11
Convolutional Integrator
12
Memory
53
22
2
16
Reverberation Circuit
14
Correcting Circuit
1
Two-Channel Analog Stereophonic Signal Source

250 A signal source (2) transmits analog two-channel audio signals, which may originate from a laser disc or an analog source, for example. These signals are converted into digital signals in two analog-to-digital converters (3) (col. 11, lines 8-14).

251 The digital signals are subjected to a convolution integral by convolution integrators (5, 7, 9, 11) equipped with memories (6, 8, 10, 12) and then corrected by control devices (50, 51, 52, 53). For this purpose, control signals are used from which the arrival time and sound pressure level can be read in response to a head rotation. Head inclinations are detected by a gyroscope (28) attached to the headphones. The signals are then forwarded to adders (15, 16) (col. 11, lines 56-66).

252 Information is stored in a memory (35) in order to detect and take head movements into account. This information may consist of impulse responses from which differences in time spans, levels, and the like between the sounds arriving at the two ears can be derived, based on the virtual positions of virtual sound sources in relation to the orientation of the ears when the listener turns his head relative to the reference orientation of the head (col. 12, line 65 to col. 13, line 6).

253 Two reverberation circuits (13, 14) generate reverberation signals that can be added to the signal to create specific spatial impressions (col. 22, lines 44-49). Adders (15, 16) are used for this purpose (col. 28, lines 52 f.).

254 The digital two-channel signals generated in this way are corrected by correction circuits (17, 18) in order to eliminate differences in the shape of the ears and the characteristics inherent in the sound sources and headphones used. The signals are then subjected to converting into analog two-channel signals by means of D/A converters (19, 20), amplified by power amplifiers (21, 22), and forwarded to the headphones (24) (col. 11, lines 63 to col. 12, lines 3).

255 The control devices (50, 51, 52, 53) can be formed by combining a variable delay device and a variable level controller and by a variable digital IIR or FIR filter (col. 12, line 61 to col. 13, line 6).

256 After A/D conversion, the digital signals can optionally be converted into the frequency domain by means of a Fourier transform. In this case, converting back into the time domain takes place before conversion into analog signals (col. 46, lines 1-13).

257 bb) Thus, features 1.6.1 and 1.6.2 are neither disclosed nor suggested.

258 Similar to the methods described in NK2 and NK3, the method described in NK12 uses input signals that are already divided into two separate channels. The processing of BCC signals is not disclosed and, for the reasons already explained in connection with NK3, is also not suggested.

259 Contrary to the plaintiff's opinion, the delay and scaling values determined by taking into account the listener's head movements using a gyroscope are not data on time and level differences between the channels within the meaning of features 1.6.1 and 1.6.2. These values are not derived from input signal data, but are calculated from additional position information derived from the user's head movements.

260 h) No further suggestions can be derived from US patent specification 5,943,427 (NK13).

261 aa) NK13 deals with digital sound generation systems that simulate the three-dimensional position of one or more emitters and the reflecting surfaces in relation to an acoustic receiver (col. 1, lines 18-21).

262 The description of NK13 explains that the human brain relies on numerous characteristics of the received sound, known as cues, to determine the position of an emitter. These include the volume of a signal, the time of its arrival at the right and left ear, or arrival delays based on reflections. These cues could be synthesized to enhance the three-dimensionality of sound reproduction, for example in video games or when listening through headphones. However, known systems were not very accurate or led to artifacts (col. 1, lines 22-52).

263 bb) To improve this, NK13 proposes a system that includes, among other things, a feedback controller for generating delays.

264 An example of implementation is shown schematically in Figure 3 below.

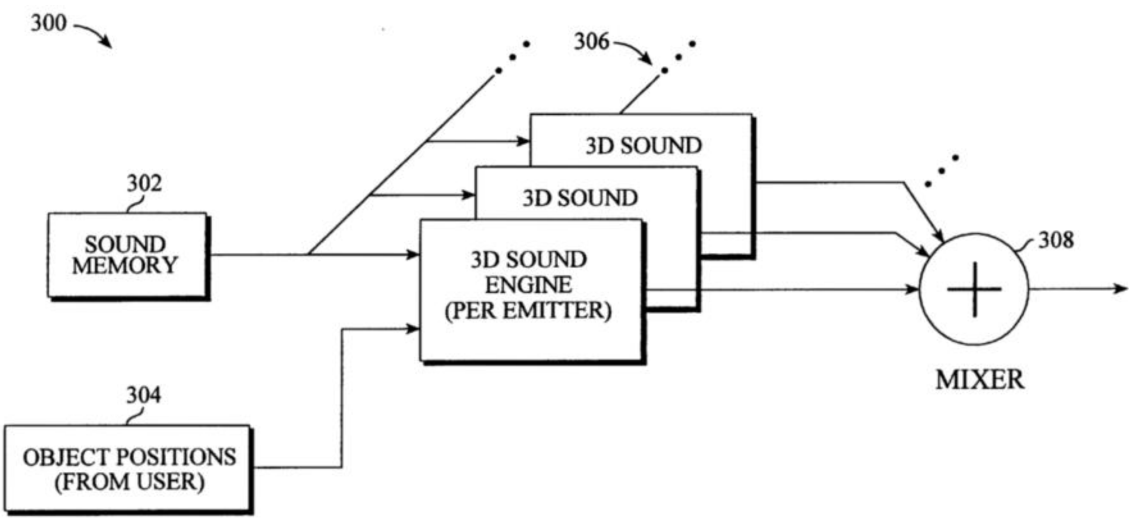

Fig. 3

265 An emitter (302) contains digital sound patterns, for example from a digital file of the known .wav format, from a microphone or from another source.

A generator (304) simulates the position and velocity of the emitter to be simulated, as well as the position and orientation of the reflective surfaces to be simulated. Three three-dimensional spatialization systems (306) each correspond to an emitter whose position or velocity is to be simulated. Since many spatialization cues require two independently positioned sound-producing devices, each spatialization system basically has separate right and left channels.

266 A mixer (308) combines the multiple emitter outputs as signals, for example for the right and left channels. If a reflective environment is to be simulated, the outputs of the individual emitters can be combined in other ways. The outputs are converted into analog signals and reproduced, for example, via headphones (col. 4, lines 21 et seq. to col. 5, line 3).

267 cc) Figure 4 below shows the structure of a three-dimensional spatialization system (306).

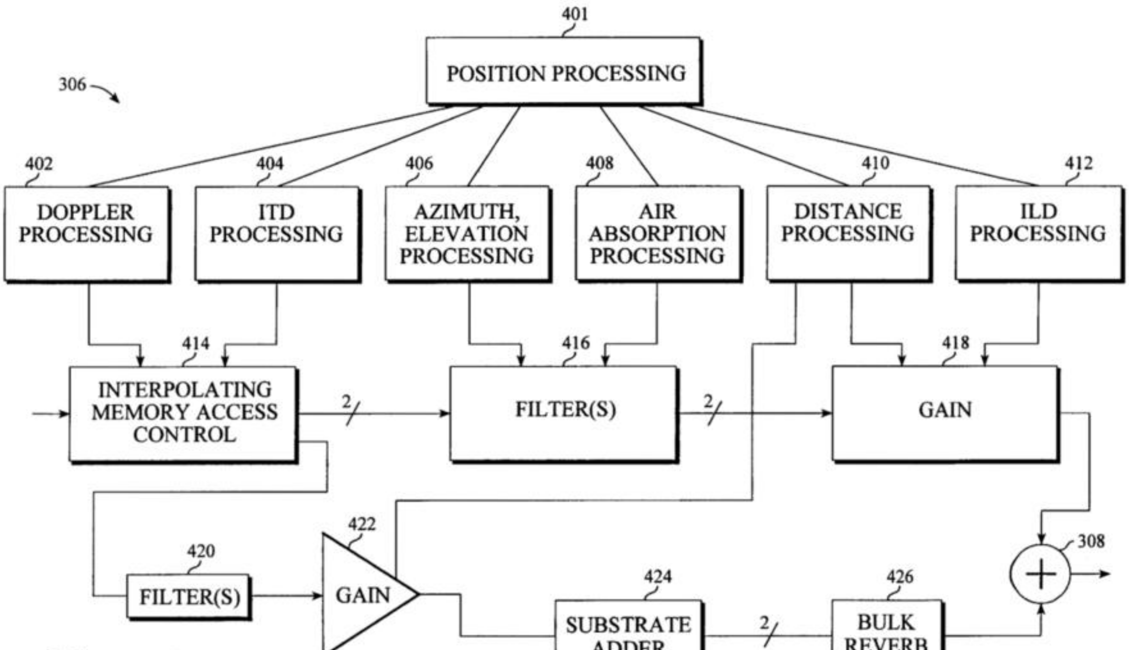

Fig. 4

268 Each spatialization system (306) comprises several subsystems that synthesize distance, velocity, and reflection cues in relation to the information

provided by the generator to define an acoustic environment.

269 An interaural time delay (ITD) processing unit (404) determines an ITD value based on the elevation and azimuth of the emitter, which simulates the difference in time delay perceived by both ears. An interaural level difference (ILD) processing unit (412) simulates the perceived difference in volume between the ears as a function of the position of the emitter.

270 The interpolating memory access controller unit (414) accesses the input of the sound memory/emitter (302) and generates undelayed amplitudes for a left and right channel, taking into account the ITD cues calculated by the ITD processing unit (404). These amplitudes are delayed using, among other things, delay values (404) from a delay memory (520, Figure 5A). A third undelayed channel amplitude is used for reverberation (col. 5, lines 27-34).

271 A first filter unit (416) is controlled by the processing units for azimuth/elevation (406) and for air absorption (408). It may comprise, for example, a comb filter and a low-pass filter (702, 704). In a preferred embodiment, the filters for the right and left channels are duplicated (col. 7, lines 60-67). The amplifier unit (418) contains separate amplifiers for the left and right channels, each with variable amplification power. The gain values are adjusted together to, among other things, create an interaural level difference cue (ILD) controlled by the interaural level difference (ILD) processing unit (412) (col. 5, lines 38-45).

272 The third undelayed channel amplitude for the reverberation is fed to the filter unit (420). This is controlled by the azimuth/elevation processing unit (406) and the air absorption processing unit (408) (col. 5, lines 46-50). It can essentially correspond to the filter unit (416). In a preferred embodiment, only a low-pass filter without a comb filter is implemented. Its parameters can be adjusted to generate a

cue that simulates the reflective properties of the reflective surfaces in the simulated acoustic environment of the listener (col. 9, lines 16-22). The output is fed into the amplification unit (422) to amplify the reverberation channel based on cues from the distance processing unit (410) (col. 5, lines 51-56). A substitution adder (424) simulates the reflections according to a respective transit time of less than 80 milliseconds. A bulk reverberation device (426) simulates the reflections according to a respective transit time of more than 80 milliseconds (col. 5, lines 57-64).

273 The outputs of the amplification unit (418) and the bulk Hall device (426) are summed (308). In a preferred embodiment, many emitters use the substitution adder (424) and the bulk Hall device (426) together (col. 6, lines 2-5).

274 dd) Even with this, features 1.6.1 and 1.6.2 are neither disclosed nor suggested.

275 Contrary to the defendant's opinion, however, NK13 discloses the generation of multiple signals from a single input signal. Values representing time or level differences between two channels are also used for this purpose. However, as in NK12, these values do not originate from data in the input signal, but from position information used to simulate the position and speed of the emitter to be simulated. Values for the time and level difference are calculated from this information and incorporated into the signal processing.

276 As the defendant rightly argues, this is not sufficient to fulfill features 1.6.1 and 1.6.2 for the reasons set out above.

277 i) Claims 7 and 8 in the version according to auxiliary request 2A have corresponding features and are therefore subject to the same assessment.

278 IV. The decision on costs is based on Section 121(2) Patent Act (PatG) as well as Sections 92(1) and 97(1) Code of Civil Procedure (ZPO).

Bacher Hoffmann Kober-Dehm

Rensen von Pückler

Previous instance:
Federal Patent Court, decision of April 17, 2023 – 5 Ni 42/20 (EP) –