670068

Section 29(1)
Regulation 3.1(2)

AUSTRALIA

PATENTS ACT 1990

NOTICE OF ENTITLEMENT

We Fraunhofer - Gesellschaft zur Forderung der angewandten Forschung EV

of Leonrodstrasse 54
80636 Munchen
GERMANY

being the applicant in respect of Application No 36254/93, state the following:-

1. The persons nominated for the grant of the patent have entitlement from the actual inventors by virtue of an Assignment therewith.

2. The persons nominated for the grant of the patent are the applicants of the application listed in the Declaration under Article 8 of the PCT.

   The basic application listed in the declaration made under Article 8 of the PCT is the first application made in a Convention country in respect of the invention.

DATED this 8th day of September,1994

FRAUNHOFER - GESELLSCHAFT ZUR FORDERUNG
DER ANGEWANDTEN FORSCHUNG EV
By its Patent Attorney
KEN MADDERN

AU9336254

(12) PATENT ABRIDGMENT (11) Document No. AU-B-36254/93
(19) AUSTRALIAN PATENT OFFICE (10) Acceptance No. 670068

(54) Title
DATA COMPRESSION PROCESS DURING STORAGE AND/OR TRANSMISSION OF DIGITAL AUDIO SIGNALS FOR STUDIO APPLICATIONS WITH PERCEPTIVE CODING AND VARIABLE LENGTH CODE

International Patent Classification(s)
(51)[5] H03M 007/40 H04B 014/02

(21) Application No. : 36254/93 (22) Application Date : 25.02.93

(87) PCT Publication Number : WO93/19536

(30) Priority Data

(31) Number 4209382 (32) Date 23.03.92 (33) Country DE GERMANY

(43) Publication Date : 21.10.93

(44) Publication Date of Accepted Application : 04.07.96

(71) Applicant(s)
FRAUNHOFER-GESELLSCHAFT ZUR FORDERUNG DER ANGEWANDTEN FORSCHUNG E.V.

(72) Inventor(s)
KARL-HEINZ BRANDENBURG; JURGEN HERRE; DIETER SEITZER

(74) Attorney or Agent
KEN MADDERN , Patent Attorney, 5th Floor 150 Grenfell St, ADELAIDE SA 5000

(56) Prior Art Documents
US 5028995
US 4656500
US 4546342

(57) Claim

1. Process for reduction in the amount of data required for storage and/or transmission of digital audio signals, in which, prior to storage and/or transmission, a digital audio signal is transformed with the aid of a filter bank from the time domain into the frequency domain and from the signal in the frequency domain and using the psychoacoustic masking effect irrelevant or redundant signal information is excluded, in which the signal is quantized and coded with a variable word length and in which the signal, after reading out from the storage and/or transmission, is decoded and transformed from the frequency domain into the time domain, characterized in that the transformation of the audio signal takes place with a filter bank having accurate reconstruction properties, that during the quantization and coding process the psychoacoustic masking effect is at most half utilised and that for the variable word length code use is made of a cascaded, multidimensional Huffman code.

OPI DATE 21/10/93 APPLN. ID 36254/93
AOJP DATE 23/12/93 PCT NUMBER PCT/DE93/00177

AU9336254

(51) Internationale Patentklassifikation 5 :

H04B 1/00, H03M 7/40

A1

(11) Internationale Veröffentlichungsnummer: **WO 93/19536**

(43) Internationales Veröffentlichungsdatum: 30. September 1993 (30.09.93)

(21) Internationales Aktenzeichen: PCT/DE93/00177

(22) Internationales Anmeldedatum: 25. Februar 1993 (25.02.93)

(30) Prioritätsdaten:
P 42 09 382.1 23. März 1992 (23.03.92) DE

(71) Anmelder *(für alle Bestimmungsstaaten ausser US)*: FRAUNHOFER-GESELLSCHAFT ZUR FÖRDERUNG DER ANGEWANDTEN FORSCHUNG E.V. [DE/DE]; Leonrodstraße 54, D-8000 München 19 (DE).

(72) Erfinder; und
(75) Erfinder/Anmelder *(nur für US)* : BRANDENBURG, Karl-Heinz [DE/DE]; Haagstraße 32, D-8520 Erlangen (DE). HERRE, Jürgen [DE/DE]; Am Röthelheim 9, D-8520 Erlangen (DE). SEITZER, Dieter [DE/DE]; Humboldtstraße 14, D-8520 Erlangen (DE).

(81) Bestimmungsstaaten: AU, CA, JP, KR, NO, RU, UA, US, europäisches Patent (AT, BE, CH, DE, DK, ES, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE).

Veröffentlicht
Mit internationalem Recherchenbericht.
Vor Ablauf der für Änderungen der Ansprüche zugelassenen Frist. Veröffentlichung wird wiederholt falls Änderungen eintreffen.

670068

(54) Title: DATA COMPRESSION PROCESS DURING STORAGE AND/OR TRANSMISSION OF DIGITAL AUDIO SIGNALS FOR STUDIO APPLICATIONS WITH PERCEPTIVE CODING AND VARIABLE LENGTH CODE

(54) Bezeichnung: VERFAHREN ZUR DATENREDUKTIOIN BEI DER SPEICHERUNG UND/ODER ÜBERTRAGUNG DIGITALER AUDIOSIGNALE FÜR STUDIOANWENDUNGEN MIT PERCEPTIVER CODIERUNG UND CODE VARIABLER LÄNGE

(57) Abstract

A data compression process is disclosed for the storage and/or transmission of digital audio signals. Known processes, such as ASPEC, allow high quality music to be transmitted in a data stream that contains less than 10 % of the information contained in the data stream used in a CD. Known compression processes do not measure up, however, to the requirements of the professional studio technology, as the subsequent processing of the audio signals that takes place in the studio leads to audible disturbances. The disclosed process is insensitive to subsequent processing, such as tandem coding, mixing or fading. To achieve this insensitivity, the transposition of the audio signal from the time domain to the frequency domain is carried out with perfect reconstruction by means of a filter bank, the psychoacoustic masking effect is utilised to an essentially less extent, and a cascade multidimensional Huffman code is used for coding variable word lengths. This process allows audio signals destined to be used in studio technology to be transmitted at a data rate of 192 kbit/sec per channel.

(57) Zusammenfassung

Beschrieben wird ein Verfahren zur Datenreduktion bei der Speicherung und/oder Übertragung digitaler Audiosignale. Bekannte Verfahren, wie z.B. ASPEC ermöglichen die Übertragung hochqualitativer Musik mit einem Datenstrom, der weniger als 10 % der Informationen des bei der CD verwendeten Datenstromes enthält. Allerdings sind bekannte Reduktionsverfahren den Anforderungen in der professionellen Studiotechnik nicht gewachsen, da die dort vorgesehene Nachbearbeitung der Audiosignale zu hörbaren Störungen führt. Das erfindungsgemäße Verfahren ist unempfindlich gegen Nachbehandlungen, wie Tandemcodierung, Mischen oder Fading. Diese Unempfindlichkeit wird dadurch erreicht, daß die Abbildung des Audiosignals aus dem Zeitbereich in den Frequenzbereich mit Hilfe einer Filterbank mit perfekter Rekonstruktion erfolgt, daß der psychoakustische Verdeckungseffekt in wesentlich geringerem Maße ausgenutzt wird, und daß als Code variabler Wortlänge ein kaskadierter mehrdimensionaler Huffmancode verwendet wird. Das Verfahren erlaubt eine Übertragung von Audiosignalen für die Studiotechnik mit einer Datenrate von 192 kbit/sec je Kanal.

PROCESS FOR DATA REDUCTION IN THE STORAGE AND/OR TRANSMISSION OF DIGITAL AUDIO SIGNALS FOR STUDIO USES

DESCRIPTION

TECHNICAL FIELD

The invention relates to a process for data reduction in the storage and/or transmission of digital audio signals according to the preamble of claim 1.

Such data reduction processes are used if, during the storage and/or transmission of audio signals, an interference-free, high sound quality is required, but not such a high transmission or storage band width as is e.g. available with a compact disk (CD).

Examples are the planned, digital, terrestrial broadcasting, the sound channel of digital video recorders or tape recorders with a stationary sound head and the like.

PRIOR ART

A process according to the preamble is e.g. known from DE-OS 39 12 605. In this process the coding of the quantized spectral values takes place with an optimum coder, in which the code word used for coding decreases in length the more frequently the spectral coefficient occurs. The code is taken from a table, whose length corresponds to the number of code words (Huffman code).

A further development of this process is described in "ASPEC Coding", Proceedings of the 10th International AES Conference, London 1991, pp 71-80. In this process optimum coding, an improved psychoacoustic model and improved measures for overcoming so-called preechos are combined.

Reference is made to the aforementioned documents for explaining terms not described in detail here.

AUSTRALIAN PATENT OFFICE
ML

It has been possible with the last-mentioned process to transmit music with bit rates of only 64 kbit/sec per channel with a CD quality, the differences between the original signal and the transmitted signal being virtually imperceptible. Therefore it is possible to fulfil the requirements of consumers in connection with most uses.

However, hitherto data reduction processes have only been usable to a limited extent for professional purposes in studios. The reason for this is that reprocessing operations such as mixing, cross-fading or multiple coding and decoding modify the signal in such a way that initially masked interference caused by the coding process become audible.

DESCRIPTION OF THE INVENTION

The problem of the invention is to so further develop a process for reducing data in accordance with the preamble that is appropriate for the requirements made in connection with professional uses in studios. This problem is solved by a process having the features of claim 1.

Further developments of the invention are characterized in the subclaims.

According to the invention the transformation of the audio signal from the time domain into the frequency domain takes place with a filter bank that accurately reconstructs the signal in the time domain, i.e. without quantization and coding, the signal transformed back into the time domain completely coincides with the input signal.

Therefore quantization exclusively contributes to a coding error. If this error can be kept in the amplitude resolution range, the reprocessing of the signal does not lead to audible interference. In order to reduce errors caused by quantization and coding, the psychoacoustic masking effect is not completely utilized.

AUSTRALIAN PATENT OFFICE

In known processes a ~~considered~~ block section is coded with such a small number of bits that noise caused by the quantization is still below the monitoring threshold, i.e. masked by the music signal. As a result of the reprocessing of the signal the initially masked quantization noise can become an audible, disturbing noise.

Thus, according to the invention, during quantization and coding the psychoacoustic masking effect, compared with the aforementioned ASPEC process, is at the most half used. Thus, there is a spacing between the masked and masking signal, which permits a reprocessing of the music signal.

In the process according to the invention coding takes place with a variable word length and for this ~~as the~~ code use is made of a cascaded, multidimensional Huffman code.

The Huffman coding is not performed in the same way as in known processes (e.g. ASPEC). Due to the significant data reduction, in known processes mainly low values occur, which must be coded. By means of a Huffman code table code words are allocated to these values. The coding of highly quantized values, which are not contained in the table, takes place with the aid of an identification code, which initiates a further coding stage.

Contrary to the above procedure, in the process according to the invention, where far more high values must be coded, coding takes place with different Huffman code tables. Firstly a decision is made as to the range in which the value occurs. Corresponding to said range the code word is associated by means of the suitable Huffman code table. If the quantized value exceeds a specific threshold, then the part exceeding said threshold is coded in a scalar manner. For small values, coding takes place in the known manner.

For calculating the masking effects in the known process use is

AUSTRALIAN ML PATENT OFFICE

made of the monitoring threshold curve (spreading function). In order to increase the ratio of masking to masked signal, according to claim 2, in place of the monitoring threshold curve obtained from the psychoacoustics, a curve with an at least twice the slope ~~double slope steepness~~ is chosen. Preferably the slopes of the monitoring threshold curve are made so steep that only a slight masking is assumed. As a result of a reprocessing such as the filtering for balancing the signal, during said coding the quantization noise does not pass into an audible range.

According to claim 3 the amplitude of the noise resulting from the quantization is at least 14 dB below the masking audio signal. This ensures that even in the case of multiple application of the process there is no audible deterioration of the music signal.

In order to obviate possible problems during a subsequent amplification of audio signals with low amplitude, according to claim 4, for the fundamental audibility threshold a level is assumed which is at least 10 dB below the amplitude resolution (least significant bit).

In a particularly advantageous development of the invention in accordance with claim 5, the filter bank is constituted by a modified discrete cosine transformation (MDCT), which according to claim 6 is used with a block length of 256 or 512 samples. For a sampling rate of 48 kHz this leads to block length of 5.33 or 10.67 msec.

MDCT is a transformation with perfect reconstruction, i.e. the output signal of the synthesis filter bank is precisely identical to the input signal of the analysis filter bank if no quantization is performed. The option to use a block length of 256 samples makes it possible to increase the time resolution in the case of reduced coding efficiency.

AUSTRALIAN ML PATENT OFFICE

According to claim 7 the analysis window has 512 or 1024 samples and is therefore 10.67 or 21.33 msec long for a sampling rate of 48 kHz. Apart from the sampling rates of 32, 44.1 and 48 kHz used in conventional processes, the present process also has a sampling rate of 96kHz. At the latter the process makes it possible to transmit frequencies well above the human perceptivity, which is sometimes a requirement of studio applications.

An advantageous further development of the invention is characterized in claim 9, according to which the coded signal is reprocessed, e.g. faded in or out. For this purpose the coded signal can be input into a computer and directly processed. This process makes it possible to bring about considerable savings in digital processing and memory allocation.

As in a sound studio, generally several music channels are mixed, according to the further development of the invention of claim 10, up to 48 channels can be processed in parallel.

An important advantage of the invention is that a digital music signal processed with the process is suitable for professional use in sound studios. The signal can be mixed and cross-faded, it can be played with widely differing volumes, it can be coded and decoded several times in succession, without any deterioration of the sound signal.

The signal can be provided with an echo or other effects. All the aforementioned measures can be used on individual channels, which can be combined to an overall signal.

Unlike data reduction processes, which do not produce noise, such as those used for the storage of computer data, the process according to the invention leads to a much greater reduction.

The process makes it possible to obtain an almost perfect reconstruction of the music signal, with only 4 to 6 bits being required per sample. Thus, compared with the original signal of 16

AUSTRALIAN PATENT OFFICE

or 20 bits per sample, a reduction factor of 4 to 5 is obtained. The typical bit rate is 192 kbit/sec in the case of an almost lossless transmission.

WAY OF PERFORMING THE INVENTION

An embodiment of the process according to the invention is described in greater detail hereinafter.

A digital audio signal is transformed from the time domain into the frequency domain by a modified discrete cosine transformation in order to break the input signal down into subsampled spectral components. The sampling rate is 48 kHz, the block length 516 samples, 8 or 10.67 msec and the window width is 1024 samples.

The output signals of this filter bank are used for calculating estimates of the particular signal-dependent monitoring thresholds. For this purpose all laws known from psychoacoustics are used. However, for determining the minimum monitoring threshold use is made of a modified acoustic model. The basis is formed by model 2 of the coder of the ISO/MPEG standard 11172-3 (International Standardisation Organisation/Moving Pictures Experts Group), but which has been modified. The gradient of the slopes of the monitoring threshold curve has been increased to such an extent that account is only taken of a minimum masking. It has proved particularly advantageous to use a monitoring threshold curve with a slope gradient, which is at least twice as steep as the curve of model 2 of the ISO/MPEG standard. The spacing between the concealing signal and the concealed noise is 20 dB instead of 6 to 9 dB for signals with a similar noise. The signal is quantized with a linear quantizer, because the latter permits an optimum ratio of the concealing signal and the concealed noise. The quantized spectral values are coded with the aid of Huffman coding. As considerably more high values have to be coded than in the known ASPEC process, the code tables are organized differently. The high values are coded in that it is established in which range the value occurs. A choice is made from the different Huffman code tables of

AUSTRALIAN PATENT OFFICE ML

that corresponding to the range. If the quantized values exceed a given threshold, then through the table scalar values are coded instead of value pairs. The coded values and side information are combined to an output bit stream.

As the psychoacoustic model is only used in the coder and not in the decoder, the model can be modified without modifying the bit stream definition.

In order to build up a very simple coder for performing the process, other acoustic models adapted to the particular requirements can be used as a basis.

AUSTRALIAN PATENT OFFICE

CLAIMS

1. Process for reduction in the amount of data required for storage and/or transmission of digital audio signals, in which, prior to storage and/or transmission, a digital audio signal is transformed with the aid of a filter bank from the time domain into the frequency domain and from the signal in the frequency domain and using the psychoacoustic masking effect irrelevant or redundant signal information is excluded, in which the signal is quantized and coded with a variable word length and in which the signal, after reading out from the storage and/or transmission, is decoded and transformed from the frequency domain into the time domain, characterized in that the transformation of the audio signal takes place with a filter bank having accurate reconstruction properties, that during the quantization and coding process the psychoacoustic masking effect is at most half utilised and that for the variable word length code use is made of a cascaded, multidimensional Huffman code.

2. Process according to claim 1, characterized in that for the calculation of masking effects in place of the monitoring threshold curve use is made of a curve with at least twice the slope.

3. Process according to claim 1 or 2, characterized in that for noisy signals use is made of a masking threshold which is at least 14 dB below the signal amplitude.

4. Process according to any one of the claims 1 to 3, characterized in that as the fundamental audibility threshold a level is assumed which is at least 10 dB below the amplitude resolution.

5. Process according to any one of the claims 1 to 4, characterized in that a modified discrete cosine transformation (MDCT) is used as the filter bank.

AUSTRALIAN PATENT OFFICE

6. Process according to claim 5, characterized in that the modified discrete cosine transformation is used with a block length of 256 or 512 samples.

7. Process according to claim 5 or 6, characterized in that there is an analysis window of 512 or 1024 samples.

8. Process according to any one of the claims 1 to 7, characterized in that the incoming audio signals are selectively sampled with a sampling rate of 32, 44.1, 48 or 96 kHz.

9. Process according to any one of the claims 1 to 8, characterized in that the coded signal is reprocessed.

10. Process according to any one of the claims 1 to 9, characterized in that up to 48 channels can be processed in parallel.

AUSTRALIAN ML PATENT OFFICE

ABSTRACT

A description is given of a process for data reduction in the storage and/or transmission of digital audio signals. Known processes such as ~~e.g.~~ ASPEC permits the transmission of high quality music with a data stream containing less than 10% of the information of the data stream used in the CD. However, known reduction processes do not meet the demands of professional studio technology, because the reprocessing of the audio signals used there leads to audible interference.

The process according to the invention is not sensitive to reprocessing operations, such as tandem coding, mixing or fading. This insensitivity is achieved in that the imaging of the audio signal from the time range into the frequency range takes place with the aid of a filter bank having accurate ~~perfect~~ reconstruction properties, in that the psychoacoustic masking effect is used to a much smaller extent and in that as the variable word length code use is made of a cascaded, multidimensional Huffman code.

The process permits a transmission of audio signals for studio technology with a data rate of 192 kbit/sec/channel.

AUSTRALIAN PATENT OFFICE
ML

INTERNATIONAL SEARCH REPORT

International application No.
PCT/DE 93/00177

A. CLASSIFICATION OF SUBJECT MATTER

IPC5 H04B 1/00, H03M 7/40

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC5 H03K 13/00, H03M 7/00, H04B 1/00

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US,A,4 546 342 (WEAVER et al.) 08 October 1985 (08.10.85) abstract; claim 1	1,10
A	DE, A1, 3 930 760 (BOSCH) 28 March 1991 (28.03.91) abstract; claim 1	1,10
A	US,A,4 656 500 (MORI) 07 April 1987 (07.04.87) abstract; columns 1,5,6; claims 1-3	1,10
A	US,A,5 028 995 (IZAWA) 02 July 1991 (02.07.91) abstract; figure 3	1,10

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

* Special categories of cited documents:
"A" document defining the general state of the art which is not considered to be of particular relevance
"E" earlier document but published on or after the international filing date
"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
"O" document referring to an oral disclosure, use, exhibition or other means
"P" document published prior to the international filing date but later than the priority date claimed
"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
27 June 1993 (27.06.93)	30 July 1993 (30.07.93)
Name and mailing address of the ISA/ EUROPEAN PATENT OFFICE	Authorized officer
Facsimile No.	Telephone No.

INTERNATIONALER RECHERCHENBERICHT

Internationales Aktenzeichen PCT/DE 93/00177

I. KLASSIFIKATION DES ANMELDUNGSGEGENSTANDS (bei mehreren Klassifikationssymbolen sind alle anzugeben)[6]

Nach der Internationalen Patentklassifikation (IPC) oder nach der nationalen Klassifikation und der IPC

Int.Cl[5] H 04 B 1/00,H 03 M 7/40

II. RECHERCHIERTE SACHGEBIETE

Recherchierter Mindestprüfstoff[7]

Klassifikationssystem	Klassifikationssymbole
Int.Cl.[5]	H 03 K 13/00,H 0: M 7/00,H 04 B 1/00

Recherchierte nicht zum Mindestprüfstoff gehörende Veröffentlichungen, soweit diese unter die recherchierten Sachgebiete fallen[8]

III. EINSCHLÄGIGE VERÖFFENTLICHUNGEN[9]

Art*	Kennzeichnung der Veröffentlichung[11], soweit erforderlich unter Angabe der maßgeblichen Teile[12]	Betr. Anspruch Nr.[13]
A	US, A, 4 546 342 (WEAVER et al.) 08 Oktober 1985 (08.10.85) Zusammenfassung; Anspruch 1. --	1,10
A	DE, A1, 3 930 760 (BOSCH) 28 März 1991 (28.03.91) Zusammenfassung; Anspruch 1. --	1,10
A	US, A, 4 656 500 (MORI) 07 April 1987 (07.04.87) Zusammenfassung; Spalten 1,5, 6; Ansprüche 1-3. --	1,10
A	US, A, 5 028 995 (IZAWA) 02 juli 1991 (02.07.91)	1,10

* Besondere Kategorien von angegebenen Veröffentlichungen[10]:

"A" Veröffentlichung, die den allgemeinen Stand der Technik definiert, aber nicht als besonders bedeutsam anzusehen ist

"E" älteres Dokument, das jedoch erst am oder nach dem internationalen Anmeldedatum veröffentlicht worden ist

"L" Veröffentlichung, die geeignet ist, einen Prioritätsanspruch zweifelhaft erscheinen zu lassen, oder durch die das Veröffentlichungsdatum einer anderen im Recherchenbericht genannten Veröffentlichung belegt werden soll oder die aus einem anderen besonderen Grund angegeben ist (wie ausgeführt)

"O" Veröffentlichung, die sich auf eine mündliche Offenbarung, eine Benutzung, eine Ausstellung oder andere Maßnahmen bezieht

"P" Veröffentlichung, die vor dem internationalen Anmeldedatum, aber nach dem beanspruchten Prioritätsdatum veröffentlicht worden ist

"T" Spätere Veröffentlichung, die nach dem internationalen Anmeldedatum oder dem Prioritätsdatum veröffentlicht worden ist und mit der Anmeldung nicht kollidiert, sondern nur zum Verständnis des der Erfindung zugrundeliegenden Prinzips oder der ihr zugrundeliegenden Theorie angegeben ist

"X" Veröffentlichung von besonderer Bedeutung; die beanspruchte Erfindung kann nicht als neu oder auf erfinderischer Tätigkeit beruhend betrachtet werden

"Y" Veröffentlichung von besonderer Bedeutung; die beanspruchte Erfindung kann nicht als auf erfinderischer Tätigkeit beruhend betrachtet werden, wenn die Veröffentlichung mit einer oder mehreren anderen Veröffentlichungen dieser Kategorie in Verbindung gebracht wird und diese Verbindung für einen Fachmann naheliegend ist

"&" Veröffentlichung, die Mitglied derselben Patentfamilie ist

IV. BESCHEINIGUNG

Datum des Abschlusses der internationalen Recherche	Absendedatum des internationalen Recherchenberichts
27 Juni 1993	30.07.93
Internationale Recherchenbehörde	**Unterschrift des bevollmächtigten Bediensteten**
Europäisches Patentamt	BLASL e.h.

Internationales Akter. ...chen PCT/DE 93/00177

III. EINSCHLÄGIGE VERÖFFENTLICHUNGEN (Fortsetzung von Blatt 2)		
Art *	Kennzeichnung der Veröffentlichung, soweit erforderlich unter Angabe der maßgeblichen Teile	Betr. Anspruch Nr.
	Zusammenfassung; Fig. 3. ----	

ANHANG

ANNEX

ANNEXE

zum internationalen Recherchenbericht über die internationale Patentanmeldung Nr.

to the International Search Report to the International Patent Application No.

au rapport de recherche international relatif à la demande de brevet international n°

PCT/DE 93/00177 SAE 71190

In diesem Anhang sind die Mitglieder der Patentfamilien der im obengenannten internationalen Recherchenbericht angeführten Patentdokumente angegeben. Diese Angaben dienen nur zur Unterrichtung und erfolgen ohne Gewähr.

This Annex lists the patent family members relating to the patent documents cited in the above-mentioned international search report. The Office is in no way liable for these particulars which are given merely for the purpose of information.

La présente annexe indique les membres de la famille de brevets relatifs aux documents de brevets cités dans le rapport de recherche international visée ci-dessus. Les reseignements fournis sont donnés à titre indicatif et n'engagent pas la responsibilité de l'Office.

Im Recherchenbericht angeführtes Patentdokument Patent document cited in search report Document de brevet cité dans le rapport de recherche	Datum der Veröffentlichung Publication date Date de publication	Mitglied(er) der Patentfamilie Patent family member(s) Membre(s) de la famille de brevets	Datum der Veröffentlichung Publication date Date de publication
US A 4546342	08-10-85	CA A1 1232357	02-02-88
		DE T 3490579	12-12-85
		GB A0 8519233	04-09-85
		GB A1 2165725	16-04-86
		GB B2 2165725	08-06-88
		JP T2 61500822	24-04-86
		NL A 8420063	01-11-85
		WO A1 8502732	20-06-85
DE A1 3930760	28-03-91	keine - none - rien	
US A 4656500	07-04-87	JP A2 59200592	13-11-84
US A 5028995	02-07-91	GB A1 2211691	05-07-89
		GB B2 2211691	22-04-92
		JP A2 1243678	28-09-89
		GB A0 8825273	30-11-88
		JP A2 1114279	02-05-89