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(71) Applicant: DOLBY LABORATORIES LICENSING CORPORATION [US/US]; 100 Potrero Avenue, San Francisco, CA 94103-4813 (US).

(72) Inventors: DAVIS, Mark, Franklin ; 1110 Manzanita Drive, Pacifica, CA 94044 (US). TODD, Craig, Campbell ; 304 Durant Way, Mill Valley, CA 94941 (US).

(74) Agents: GALLAGHER, Thomas, A. et al.; The Law Offices of Thomas A. Gallagher, 100 Green Street, Third Floor, San Francisco, CA 94111-1302 (US).

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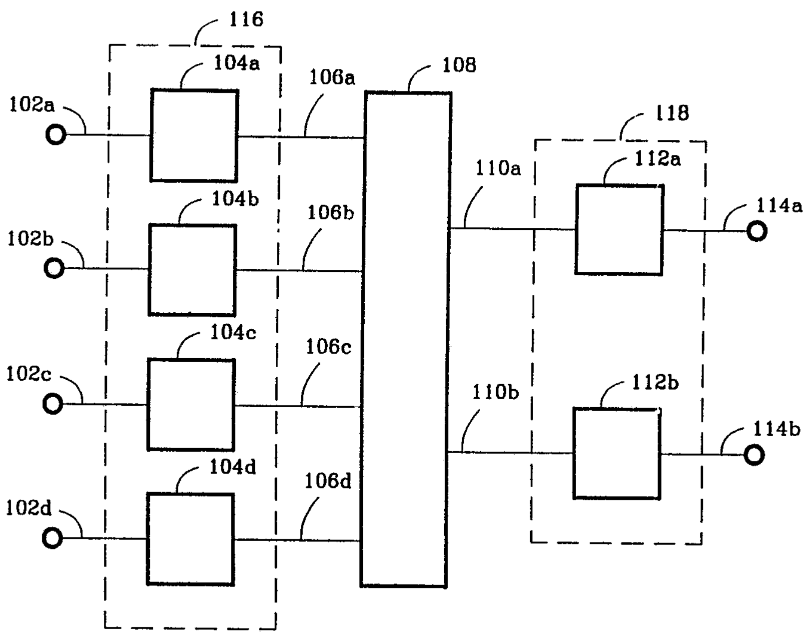

(57) Abstract

The invention relates in general to the reproducing of multi-channel signals. More particularly, the invention relates to the decoding of multi-channel audio signals representing multidimensional sound fields delivered by one or more delivery channels, wherein the complexity of the decoding is roughly proportional to the number of channels used to present the decoded signal which may differ from the number of delivery channels.

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DESCRIPTION

DECODER FOR VARIABLE-NUMBER OF CHANNEL PRESENTATION OF MULTIDIMENSIONAL SOUND FIELDS

Technical Field

The invention relates in general to the reproducing of multi-channel signals. More particularly, the invention relates to the decoding of multi-channel audio signals representing multidimensional sound fields delivered by one or more delivery channels, wherein the complexity of the decoding is roughly proportional to the number of channels used to present the decoded signal which may differ from the number of delivery channels.

Background Art

A goal for high-fidelity reproduction of recorded or transmitted sounds is the presentation at another time or location as faithful a representation of an "original" sound field as possible given the limitations of the presentation or reproduction system. A sound field is defined as a collection of sound pressures which are a function of time and space. Thus, high-fidelity reproduction attempts to recreate the acoustic pressures which existed in the original sound field in a region about a listener.

Ideally, differences between the original sound field and the reproduced sound field are inaudible, or if not inaudible at least relatively unnoticeable to most listeners. Two general measures of fidelity are "sound quality" and "sound field localization."

Sound quality includes characteristics of reproduction such as frequency range (bandwidth), accuracy of relative amplitude levels throughout the frequency range (timbre), range of sound amplitude level (dynamic range), accuracy of harmonic amplitude and phase (distortion level), and amplitude level and frequency of spurious sounds and artifacts not present in the original sound (noise). Although most aspects of sound quality are susceptible to measurement by instruments, in practical systems characteristics of the human hearing system (psychoacoustic effects) render inaudible or relatively unnoticeable certain measurable deviations from the "original" sounds.

Sound field localization is one measure of spatial fidelity. The preservation of the apparent direction (both azimuth and elevation) and distance of a sound source is sometimes known as angular and depth localization, respectively. In the case of certain orchestral and other recordings, such localization is intended to convey to the listener the actual physical placement of the musicians and their instruments. With respect to other recordings, particularly multiple track recordings produced in a studio, the angular directionality and depth may bear no relationship to any "real-life" arrangement of sound sources and the localization is merely a part of the overall artistic impression intended to be conveyed to the listener. For example, speech seeming to originate from a specific point in space may be added to a pre-recorded sound field. In any case, one purpose of high-fidelity multi-channel reproduction systems is to reproduce spatial aspects of an on-going sound field, whether real or synthesized. As with respect to sound quality, in practical systems measurable changes in localization are, under certain conditions, inaudible or relatively unnoticeable because of characteristics of human hearing.

It is sufficient to recognize that a sound-field producer may develop recorded or transmitted signals which, in conjunction with a reproduction system, will present to a human listener a sound field possessing specific characteristics in sound quality and sound field localization. The sound field presented to the listener may closely approximate the ideal sound field intended by the producer or it may deviate from it depending on many factors including the reproduction equipment and acoustic reproduction environment.

A sound field captured for transmission or reproduction is usually represented at some point by one or more electrical signals. Such signals usually constitute one or more channels at the point of sound field capture ("capture channels"), at the point of sound field transmission or recording ("transmission channels"), and at the point of sound field presentation ("presentation channels"). Although within some limits as the number of these sound channels increases, the ability to reproduce complex sound fields increases, practical considerations impose limits on the number of such channels.

In most, if not all cases, the sound field producer works in a relatively well defined system in which there are known presentation channel configurations and environments. For example, a two-channel stereophonic recording is generally expected to be presented through either two presentation channels ("stereophonic") or one presentation channel ("monophonic"). The recording is usually optimized to sound good to most listeners having either stereophonic or monophonic playback equipment. As another example, a multiple-channel recording in stereo with surround sound for motion pictures is made with the expectation that motion picture theaters will have either a known, generally standardized arrangement for presenting the left, center, right, bass and surround channels or, alternatively, a classic "Academy" monophonic playback. Such recordings are also made with the expectation that they will be played by home playback equipment ranging from single presentation-channel systems such as a small loudspeaker in a television set to relatively sophisticated multiple presentation-channel surround-sound systems.

Various techniques are sometimes used to reduce the number of transmission channels required to carry signals representing multiple-dimensional sound fields. One example of such a technique is a 4-2-4 matrix system which combines four channels into two transmission channels for transmission or storage, from which four presentation channels are extracted for playback. Ideally, such techniques should not create audible changes in the sound field when presented.

Such techniques may be used without departing from the scope of the present invention; however, it may not always be desirable to do so. The use of these techniques make it necessary to develop the concept of a "delivery channel." A delivery channel represents a discrete encoder channel, or a set of information which is independently encoded. A delivery channel corresponds to a transmission channel in systems which do not use techniques to reduce the number of transmission channels. For example, a 4-2-4 matrix system carries four delivery channels over two transmission channels, ostensibly for playback using four presentation channels. The present invention is directed toward selecting a number of presentation channels which differs from the number of delivery channels.

An example of a simple technique which generates one presentation channel in response to two delivery channels is the summing of two delivery channels to form one presentation channel. If a signal is sampled and digitally encoded using Pulse Code Modulation (PCM), the summation of two delivery channels may be performed in the digital domain by adding PCM samples representing each channel and converting the summed samples into an analog signal using a digital-to-analog converter (DAC). The summation of two PCM coded signals may also be performed in the analog domain by converting the PCM samples for each delivery channel into an analog signal using two DACs and summing the two analog signals. Performing the summation in the digital domain is usually preferred because a digital adder is generally more accurate and less expensive to implement than using a second high-precision DAC.

This technique becomes much more complex, however, if signal samples are digitally encoded in a nonlinear form rather than encoded in linear PCM. Nonlinear forms may be generated by encoding methods such as logarithmic quantizing, normalizing floating-point representations, and adaptively allocating bits to represent each sample.

Nonlinear representations are frequently used in encoder/decoder systems to reduce the amount of information required to represent the coded signal. Such representations may be conveyed by transmission channels with reduced informational capacity, such as lower bandwidth or noisy transmission paths, or by recording media with lower storage capacity.

Nonlinear representations need not reduce informational requirements. Various forms of information packing may be used only to facilitate transmission error detection and correction. The broader terms "formatted" and "formatting" will be used herein to refer to nonlinear representations and to obtaining such representations, respectively. The terms "deformatted" and "deformatting" will refer to reconstructed linear representations and to obtaining such reconstructed linear representations, respectively.

It should be mentioned that what constitutes a "linear" representation depends upon the signal processing methods employed. For example, floating-point representation is linear for a Digital Signal Processor (DSP) which can perform arithmetic with floating-point operands, but such representation is not linear for a DSP which can only perform integer arithmetic. The significance of "linear" will be discussed further in connection with the Modes for Carrying Out the Invention, below.

A decoder should use deformatting techniques inverse to the formatting techniques used to format the information to obtain a representation like PCM which can be summed as described above.

Two encoding techniques which utilize formatting to reduce informational requirements are subband coding and transform coding. Subband and transform coders attempt to reduce the amount of information transmitted in particular frequency bands where the resulting coding inaccuracy or coding noise is psychoacoustically masked by neighboring spectral components. Psychoacoustic masking effects usually may be more efficiently exploited if the bandwidth of the frequency bands are chosen commensurate with the bandwidths of the human ear's "critical bands." See generally, the Audio Engineering Handbook, K. Blair Benson ed., McGraw-Hill, San Francisco, 1988, pages 1.40-1.42 and 4.8-4.10. Throughout the following discussion, the term "subband" shall refer to portions of the useful signal bandwidth, whether implemented by a true subband coder, a transform coder, or other technique. The term "subband coder" shall refer to true subband coders, transform coders, and other coding techniques which operate upon such "subbands."

Signals in a formatted form cannot be summed directly; therefore each of the two delivery channels must be decoded before they can be combined by summation. Generally, decoding techniques such as subband decoding are relatively expensive to implement. Therefore, monophonic presentation of a two-channel signal is approximately twice as costly as monophonic presentation of a one-channel signal. The cost is approximately double because an expensive decoder is needed for each delivery channel.

One prior art technique which avoids burdening the cost of monophonic presentation of two-channel signals is matrixing. It is important to distinguish matrixing used to reduce the number presentation channels from matrixing used to reduce the number of transmission channels. Although they are mathematically similar, each technique is directed to very different aspects of signal transmission and reproduction.

One simple example of matrixing encodes two channels, A and B, into SUM and DIFFERENCE delivery channels according to

$$SUM = A + B, \text{ and}$$

$$DIFFERENCE = A - B.$$

For two-channel stereophonic playback, a presentation system can obtain the original two-channel signal by using two decoders to decode each delivery channel and de-matrixing the decoded channels according to

$$A' = \tfrac{1}{2} \cdot (SUM + DIFFERENCE), \text{ and}$$

$$B' = \tfrac{1}{2} \cdot (SUM - DIFFERENCE).$$

The notation A' and B' is used to represent the fact that in practical systems, the signals recovered by de-matrixing generally do not exactly correspond to the original matrixed signals.

For monophonic playback, a presentation system can obtain a summation of the original two-channel signal by using only one decoder to decode the SUM delivery channel.

Although matrixing solves the problem of disproportionate cost for monophonic presentation of two delivery channels, it suffers from what may be perceived as cross-channel noise modulation when it is used in conjunction with encoding techniques which reduce the informational requirements of the encoded signal. For example, "companding" may be used for analog signals, and various bit-rate reduction methods may be used for digital signals. The application of such techniques stimulates noise in the output signal of the decoder. The intent and expectation is that this noise is masked by the audio signal which stimulated it, and therefore is inaudible. When such techniques are applied to matrixed signals, the de-matrixed signal may be incapable of masking the noise.

Assume that a matrix encoder encodes channels A and B where only channel B contains an audio signal. Normally, noise is injected into the SUM and DIFFERENCE channels when the SUM and DIFFERENCE signals are coded for transmission with an analog compander or a digital bit-rate reduction technique. During decoding, the A' presentation channel will be obtained from the sum of the SUM and DIFFERENCE delivery channels. Although the A' presentation channel will not contain any audio signal, it will contain the sum of the analog modulation noise or the digital coding noise independently injected into each of the SUM and DIFFERENCE delivery channels. The A' presentation channel will not contain any audio signal to psychoacoustically mask the noise. Furthermore, the noise in channel A' may not be masked by the audio signal in channel B' because the ear can usually discern noise from audio signals, especially when the noise and the signal have different angular localization.

Techniques used to control the number of presentation channels become even more of a problem when more than two delivery channels are involved. For example, motion picture soundtracks typically contain four channels: Left, Center, Right, and Surround. Some current proposals for future motion picture and advanced television applications suggest five channels plus a sixth limited bandwidth subwoofer channel. When multiple-channel signals in a formatted form are delivered to consumers for playback on monophonic and two-channel home equipment, the question arises how to economically obtain a signal suitable for one- and two-channel presentation while avoiding the cross-channel noise modulation effect described above.

Disclosure of Invention

It is an object of the present invention to provide for the decoding of one or more delivery channels of signals encoded to represent in a formatted form a multi-dimensional sound field without artifacts perceived as cross-channel noise modulation, wherein the complexity or cost of the decoding is roughly proportional to the number of presentation channels. Although a decoder embodying the present invention may be implemented using analog or digital techniques or even a hybrid arrangement of such techniques, the invention is more conveniently implemented using digital techniques and the preferred embodiments disclosed herein are digital implementations.

In accordance with the teachings of the present invention, in one embodiment, a transform decoder receives an encoded signal in a formatted form comprising one or more delivery channels. A deformatted representation is generated for each delivery channel. Each channel of deformatted information is distributed to one or more inverse transforms for output signal synthesis, one inverse transform for each presentation channel.

It should be understood that although the use of subbands with bandwidths commensurate with the human ear's critical bandwidths allows greater exploitation of psychoacoustic effects, application of the teachings of the present invention are not so limited. It will be obvious to those skilled in the art that these teachings may be applied to wideband signals as well; therefore, reference to subbands throughout the remaining discussion should be construed as one or more frequency bands spanning the total useful bandwidth of input signals.

As discussed above, the present invention applies to subband coders implemented by any of several techniques. A preferred implementation uses a transform, more particularly a time-domain to frequency-domain transform according to the Time Domain Aliasing Cancellation (TDAC) technique. See Princen and Bradley, "Analysis/Synthesis Filter Bank Design Based on Time Domain Aliasing Cancellation," IEEE Trans. on Acoust., Speech, Signal Proc., vol. ASSP-34, 1986, pp. 1153-1161. An example of a transform encoder/decoder system utilizing a TDAC transform is provided in International Patent Application Publication Number WO 90/09022, published August 9, 1990.

The various features of the invention and its preferred embodiments are set forth in greater detail in the following Modes for Carrying Out the Invention and in the accompanying drawings.

Brief Description of Drawings

Figure 1 is a functional block diagram illustrating the basic structure of one embodiment incorporating the invention distributing four delivery channels into two presentation channels.

Figure 2 is a functional block diagram illustrating the basic structure of a single-channel subband decoder.

Figure 3 is a functional block diagram illustrating the basic structure of a multiple-channel subband decoder distributing four decoded delivery channels into two presentation channels.

Figure 4 is a functional block diagram illustrating the basic structure of one embodiment incorporating the invention distributing four delivery channels into one presentation channel.

Modes for Carrying Out the Invention

Figure 2 illustrates the basic structure of a typical single-channel subband decoder 200. Encoded subband signals received from delivery channel 202 are deformatted into linear form by deformatter 204, and synthesizer 206 generates along presentation channel 208 a full-bandwidth representation of the received signal. It should be appreciated that a practical implementation of a decoder may incorporate additional features such as a buffer for delivery channel 202, and a digital-to-analog converter and a low-pass filter for presentation channel 208, which are not shown.

As briefly mentioned above, deformatter 204 should obtain a linear representation using a method inverse to that used by a companion encoder which generated the nonlinear representation. In a practical embodiment, such nonlinear representations are generally used to reduce the informational requirements imposed upon transmission channels and storage media. Deformatting generally involves simple operations which can be performed relatively quickly and are relatively inexpensive to implement.

Synthesizer 206 represents a synthesis filter bank for true digital subband decoders, and represents an inverse transform for digital transform decoders. Signal synthesis for either type of decoder is computationally intensive, requiring many complex operations. Thus, synthesizer 206 typically requires much more time to perform and incurs much higher costs to implement than that required by deformatter 204.

Figure 3 illustrates the basic structure of a typical decoder which receives and decodes four delivery channels for presentation by two presentation channels. The encoded signal received from each of the delivery channels 302a-302d is passed through a respective one of decoders 300a-300d, each comprising a respective one of deformatters 304a-304d and a respective one of synthesizers 306a-306d, respectively. The synthesized signal is passed from each decoder along a respective one of paths 308a-308d to distributor 310 which combines the four synthesized channels into two presentation channels 312a and 312b. Distributor 310 generally involves simple operations which can be performed relatively quickly using implementations that are relatively inexpensive to implement.

Most of the cost required to implement the decoder illustrated in Figure 3 is represented by the synthesizers. The number of synthesizers is equal to the number of delivery channels; thus, the cost of implementation is roughly proportional to the number of delivery channels.

Signal synthesis is linear if, ignoring small arithmetic round-off errors, signals combined before synthesis will produce the same output signal as that produced by combining signals after synthesis. Synthesis is linear for many implementations of decoders; therefore, it is often possible to put a distributor between the deformatters and the synthesizers of such a multiple-channel decoder. Such a structure is discussed more fully below and is illustrated in Figure 1. In this manner, the cost of implementation is roughly proportional to the number of presentation channels. This is highly desirable in applications such as those proposed for advanced television systems which may receive five delivery channels, but which will provide only one or two presentation channels.

In this context, it is possible to better appreciate the meaning of the term "linear" discussed above. Briefly, any representation is considered linear if it satisfies two criteria: (1) it can be direct input for the synthesizer, and (2) it permits directly forming linear combinations such as addition or subtraction which satisfy the signal synthesis linearity property described above.

Figure 1 illustrates one embodiment of a decoder according to the present invention which forms two presentation channels from four delivery channels. The decoder receives coded information from four delivery channels 102a-102d which it deformats using deformatters 104a-104d, one for each delivery channel. Distributor 108 combines the deformatted signals received from paths 106a-106d into two signals which it passes along paths 110a and 110b to synthesizers 112a and 112b, respectively. Each of the synthesizers generates a signal which it passes along a respective one of presentation channels 114a and 114b.

One skilled in the art should readily appreciate that the present invention may be applied to a wide variety of true subband and transform decoder implementations. Details of implementation for deformatters and synthesizers are beyond the scope of this discussion; however, one may obtain details of implementation by referring to any of several International Patent Applications: Publication No. WO 90/09022 published August 9, 1990, Publication No. WO 90/09064 published August 9, 1990, and Publication No. WO 91/16769 published October 31, 1991.

One embodiment of a transform decoder according to the present invention comprises deformatters and synthesizers substantially similar to those described in Publication No. WO 90/09022. According to this embodiment, referring to Figure 1, a serial bit stream comprising frequency-domain transform coefficients grouped into subbands is received from each of the delivery channels 102a-102d. Each deformatter 104a-104d buffers the bit stream into blocks of information, establishes the number of bits adaptively allocated to each frequency-domain transform coefficient by the encoder of the bit stream, and reconstructs a linear representation for each frequency-domain transform coefficient. Distributor 108 receives the linearized frequency-domain transform coefficients from paths 106a-106d, combines them as appropriate, and distributes frequency-domain information among the paths 110a and 110b. Each of the synthesizers 112a and 112b generates time-domain samples in response to the frequency-domain information received from paths 110a and 110b by applying an Inverse Fast Fourier Transform which implements the inverse TDAC transform mentioned above. Although no subsequent features are shown in Figure 1, the time-domain samples are passed along presentation channels 114a and 114b, buffered and combined to form a time-domain representation of the original coded signal, and subsequently converted from digital form to analog form by a DAC.

Assuming that the four delivery channels 102a-102d in Figure 1 represent the left (L), center (C), right (R), and surround (S) channels of a four-channel audio system, a typical embodiment of distributor 108 combines these channels to form a two-channel stereophonic representation as follows:

$$L' = L + .7071 \cdot C + .5 \cdot S \qquad (1)$$

$$R' = R + .7071 \cdot C + .5 \cdot S \tag{2}$$

where L' = left presentation channel, and

R' = right presentation channel.

For a transform decoder, these combinations represent the summation of transform coefficients in the frequency-domain. It is understood that normally only coefficients representing the same range of spectral frequencies are combined. For example, suppose each delivery channel carries a frequency-domain representation of a 20 kHz bandwidth signal transformed by a 256-point transform. Frequency-domain transform coefficient X(0) for each delivery channel represents the spectral energy of the encoded signal carried by the respective delivery channel centered about 0 Hz, and coefficient X(1) for each delivery channel represents the spectral energy of the encoded signal for the respective delivery channel centered about 78.1 Hz (20 kHz / 256). Thus, coefficient X(1) for the L' presentation channel is formed from the weighted sum of the X(1) coefficients from each delivery channel according to equation 1. Equations 1 and 2 may be rewritten as

$$X(i)_{L'} = X(i)_L + .7071 \cdot X(i)_C + .5 \cdot X(i)_S \tag{3}$$

$$X(i)_{R'} = X(i)_R + .7071 \cdot X(i)_C + .5 \cdot X(i)_S \tag{4}$$

where $X(i)_Z$ = transform coefficient i for channel Z.

For a true subband decoder, these combinations represent the summation of corresponding time-domain samples in each subband. Thus, equations 1 and 2 may be rewritten as

$$x_j(nt)_{L'} = x_j(nt)_L + .7071 \cdot x_j(nt)_C + .5 \cdot x_j(nt)_S \tag{5}$$

$$x_j(nt)_{R'} = x_j(nt)_R + .7071 \cdot x_j(nt)_C + .5 \cdot x_j(nt)_S \tag{6}$$

where $x_j(nt)_Z$ = signal sample at time nt in subband j of channel Z.

Figure 4 represents an application of the present invention used to form one presentation channel 414 from four delivery channels 402a-402d. A typical combinatorial equation for distributor 408 in this application is

$$M' = .7071 \cdot L + C + .7071 \cdot R + S \tag{7}$$

where M' = monophonic presentation channel.

The precise forms of the combinations provided by the distributor will vary according to the application.

Although it is envisioned that the present invention will normally be used to obtain a fewer number of presentation channels than there are delivery channels, the invention is not so limited. The number of presentation channels may be the same or greater than the number of delivery channels, utilizing the distributor to prepare presentation channels according to the needs of a desired application.

For example, in the transform decoder embodiment described above, two presentation channels might be formed from one delivery channel by distributing specific frequency-domain transform coefficients to a particular presentation channel, or by randomly distributing the coefficients to either or both of the presentation channels. In embodiments using transforms which pass the phase of the spectral components, distribution may be based upon the phase. Many other possibilities will be apparent.

CLAIMS

1. A decoder of one or more delivery channels of formatted information, comprising

deformatting means for generating a deformatted representation of said one or more delivery channels of formatted information,

distribution means responsive to said deformatting means for generating one or more intermediate signals, and

synthesis means responsive to said distribution means for generating one or more presentation signals.

2. A decoder according to claim 1 wherein said deformatted representation has higher informational capacity requirements than said one or more delivery channels of formatted information.

3. A decoder according to claim 1 or 2 wherein said synthesis means applies an inverse frequency-domain to time-domain transform to each of said one or more intermediate signals.

4. A decoder according to claim 1 or 2 wherein said synthesis means applies a true subband synthesis filter bank to each of said one or more intermediate signals.

5. A decoder according to any of claims 1 through 4 wherein each of said one or more intermediate signals includes a deformatted representation of at least one of said one or more delivery channels of formatted information.

6. A decoder according to any of claims 1 through 4 wherein a deformatted representation of each of said one or more delivery channels of formatted information is included in at least one of said one or more intermediate signals.

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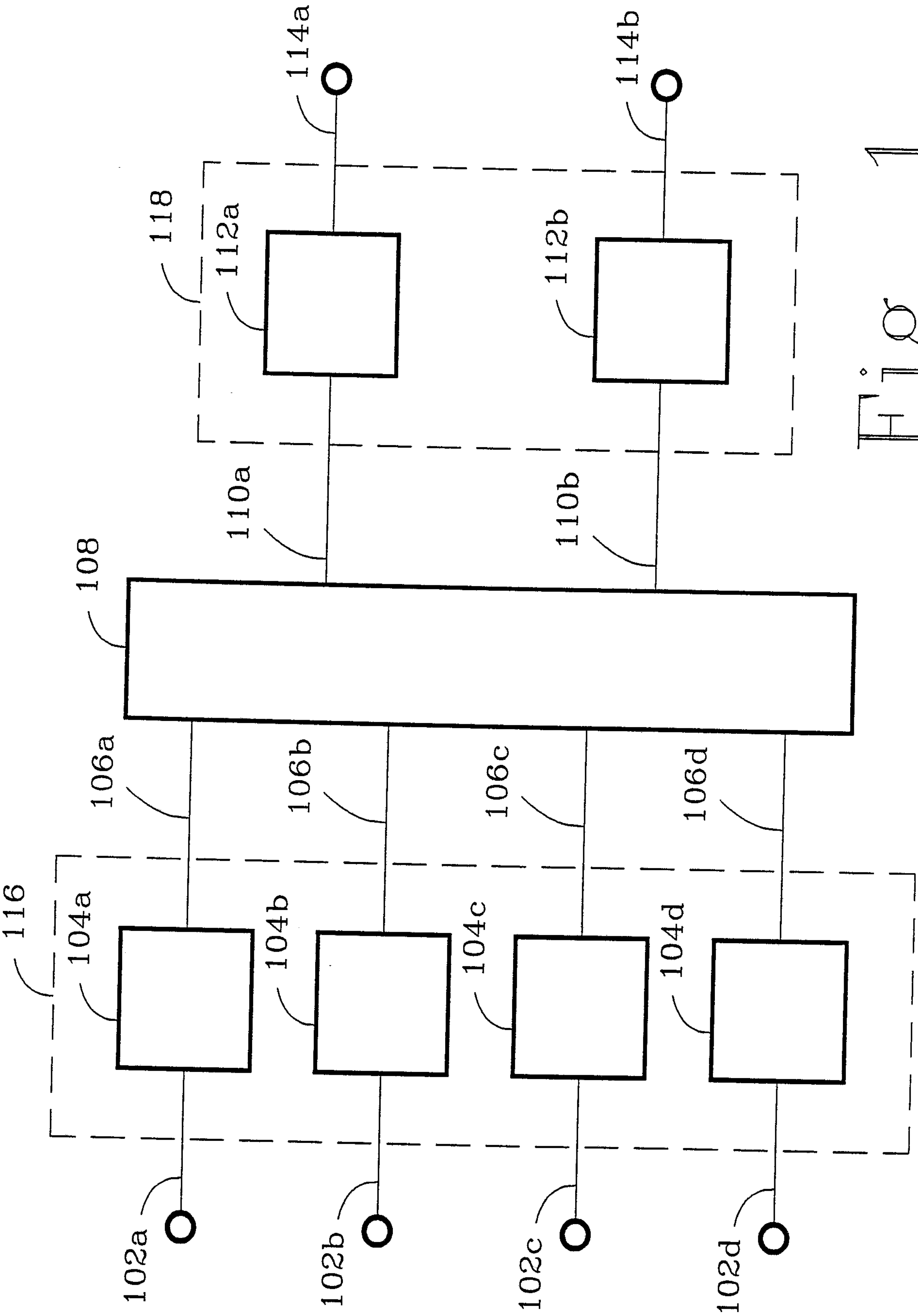
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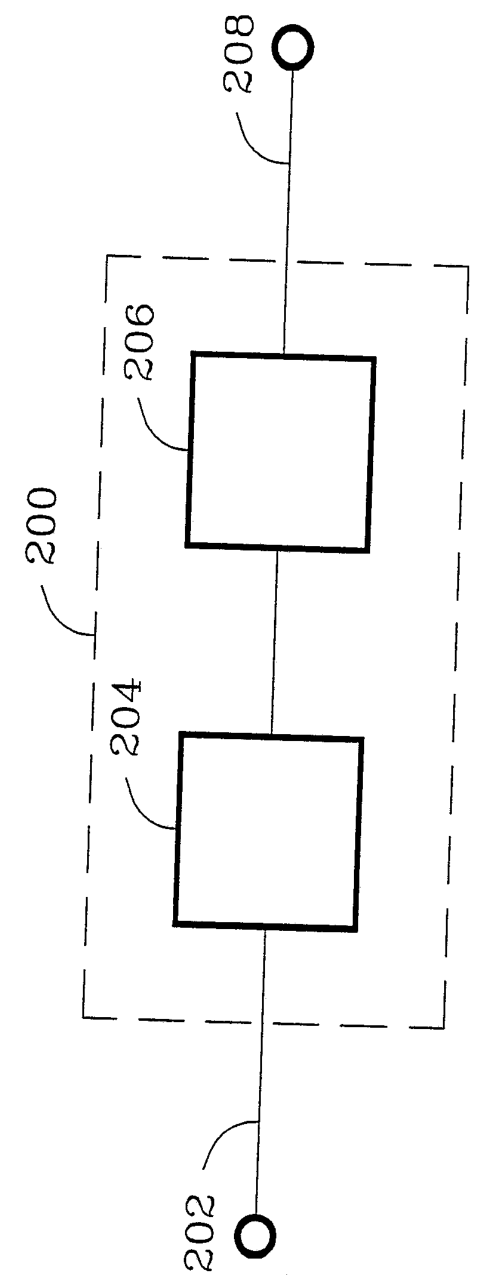
Fig. 1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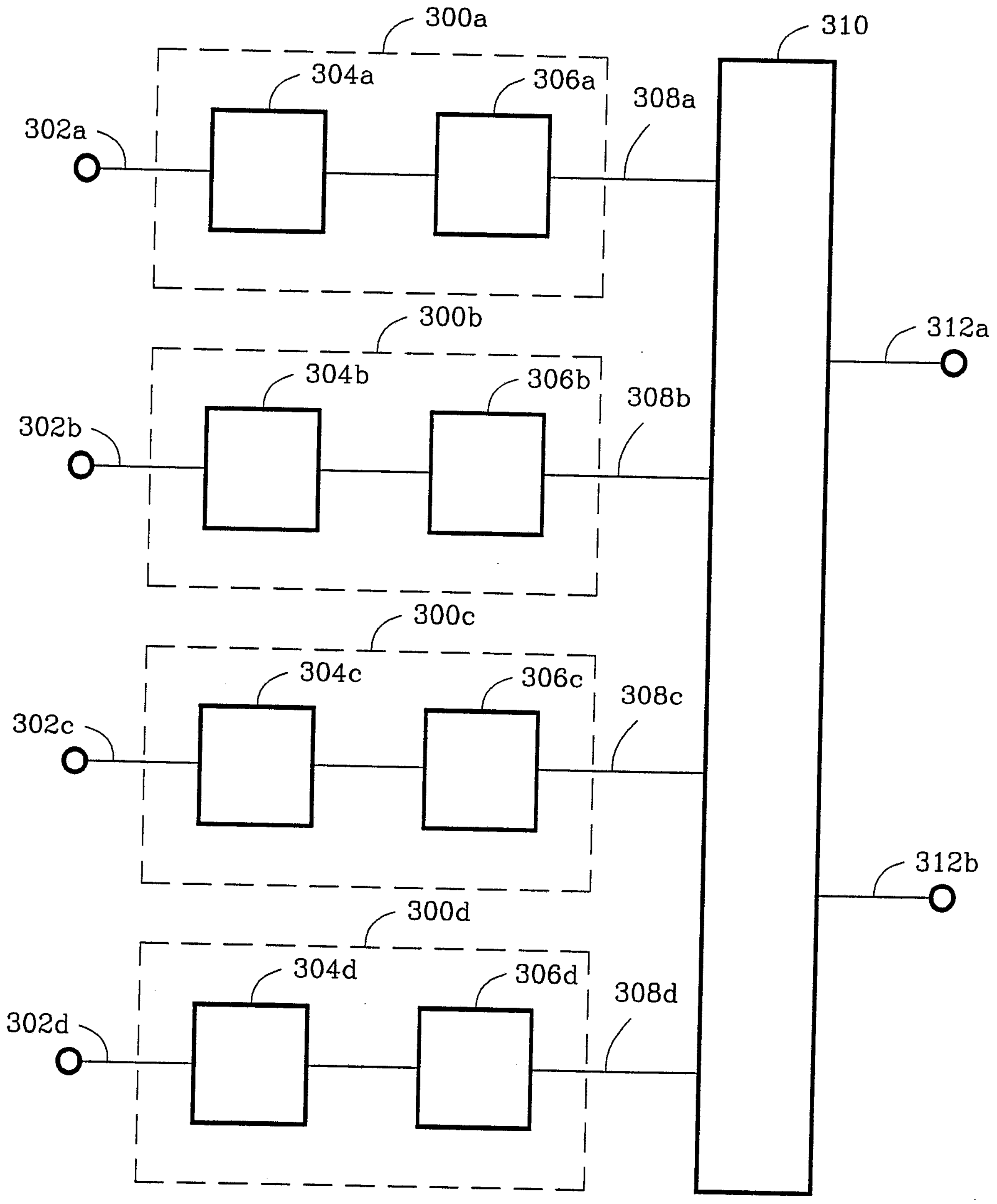
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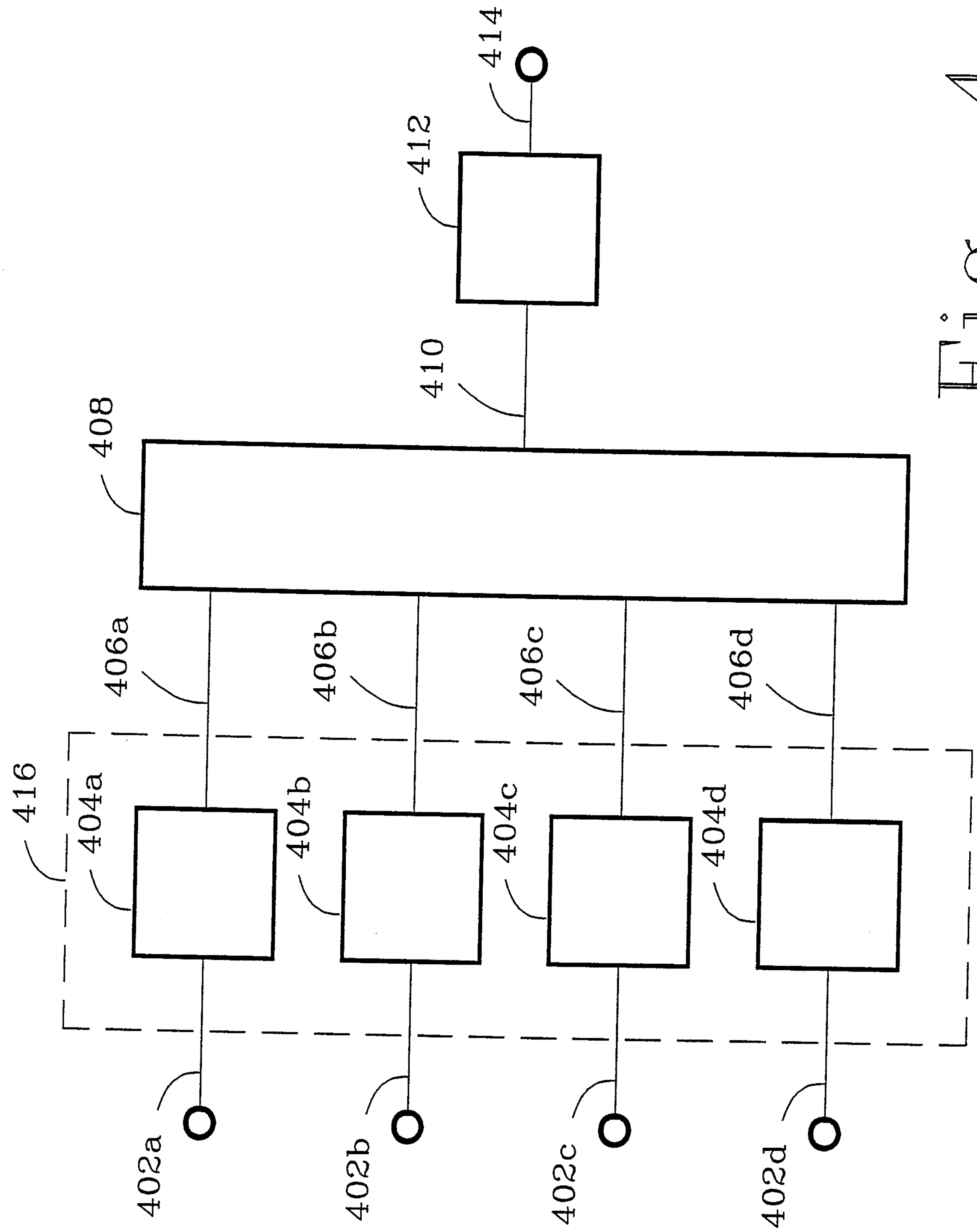
Fig. 3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402d

Fig. 4

INTERNATIONAL SEARCH REPORT

International Application No PCT/US 92/00134

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all)[6]

According to International Patent Classification (IPC) or to both National Classification and IPC

Int.Cl. 5 H04S3/00; H04S1/00; H04B1/66

II. FIELDS SEARCHED

Minimum Documentation Searched[7]

Classification System	Classification Symbols
Int.Cl. 5	H04S ; H04B ; H04H

Documentation Searched other than Minimum Documentation to the Extent that such Documents are Included in the Fields Searched[8]

III. DOCUMENTS CONSIDERED TO BE RELEVANT[9]

Category °	Citation of Document,[11] with indication, where appropriate, of the relevant passages[12]	Relevant to Claim No.[13]
X A	EP,A,0 372 601 (N.V. PHILIPS' GLOEILAMPENFABRIEKEN) 13 June 1990 see claims; figures	1,2 3-6
X	EP,A,0 402 973 (N.V. PHILIPS' GLOEILAMPENFABRIEKEN) 19 December 1990 see column 20, line 52 - column 21, line 50; figure 15	1
X	WO,A,9 016 136 (BRITISH TELECOMMUNICATIONS PUBLIC LIMITED COMPANY) 27 December 1990 see page 8, line 22 - page 9, line 14	1

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° Special categories of cited documents :[10]

"A" document defining the general state of the art which is not considered to be of particular relevance
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"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

IV. CERTIFICATION

Date of the Actual Completion of the International Search	Date of Mailing of this International Search Report
22 MAY 1992	03 JUN 1992
International Searching Authority	**Signature of Authorized Officer**
EUROPEAN PATENT OFFICE	GASTALDI G.L.

3

III. DOCUMENTS CONSIDERED TO BE RELEVANT (CONTINUED FROM THE SECOND SHEET)		
Category °	Citation of Document, with indication, where appropriate, of the relevant passages	Relevant to Claim No.
A	ICASSP 90, INT. CONF. ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING, Albuquerque, NEW MEX, USA April 3-6, 1990, vol.3, pages 1097-1100 , W.R.Th. Ten Kate et al. : ' Digital Audio Carrying Extra Information ' see the whole document ---	1

ANNEX TO THE INTERNATIONAL SEARCH REPORT ON INTERNATIONAL PATENT APPLICATION NO. US 9200134 SA 56005

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