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Error protection in a subband coder

The invention relates to a method of transmitting digital audio signals in accordance with the first part of claim 1.

From DE-OS 3506912, DE-OS - 3621513 and DE- 3639753, it is known to subject sampled, digitalised audio signals to a partial-band coding - hereinafter referred to as a subband coding. Thereby for example, the audio signals are divided up into time segments - so-called windows - corresponding to 1152 sampling values (24 msec at a sampling frequency of 48 kHz). After the break-down into 27 subbands for example, up to three scaling factors are specified for each subband. These scaling factors are quantised using e.g. 6 bits and are transmitted at a specific position in a known data frame. Their number and significance is likewise transmitted at a specific position in the data frame as an item of scaling factor selection information (Scale Factor Select Information).

Each individual data frame, as part of a bit stream divided up into sequential data frames, normally comprises a head part - a so-called header -, optionally, an error checking part - a so-called "Error Check" - , an audio data part and an ancillary data part - so-called "Ancillary Data".

After being scaled with the help of the scaling factors, the sampling values in each subband are quantised using a predetermined number of bits and transmitted at a specific position of the data frame preferably in the audio data part.

The number of bits used is likewise transmitted in coded form at a specific position in the audio data part of the data frame as a bit allocation table (Bit allocation).

A header, which includes basic items of information relating to the audio signal and the coding (e.g. stereo/mono, sampling frequency, pre-emphasis, data rate) forms the beginning of the data frame.

Furthermore, there is also a provision for optionally making some bits available in the ancillary data part at the end of the frame for ancillary items of information relating to the pieces of music (e.g. an indication of the source, the title of the music, text).

The number of bits which are to be used per subband for the coding of the sampling values is determined in a parallel operation with the aid of psycho-acoustic factors. Thereby, the total number of bits from each of the subbands has an upper limit in that each data frame can only contain a specific number of

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bits in accordance with the desired data transmission rate (e.g. 6144 bits at 256 kbits/sec). The bits utilised for the other data fields within the frame still need to be subtracted from this maximum number. This leads to the condition that there is virtually no occasion when all of the bits of the data frame contain meaningful items of information. There always remains a series of meaningless filler bits after the bits used for the sampling values. By making appropriate assumptions when calculating the number of bits which can in toto be assigned, one can control the number of filler bits, in particular by keeping it greater than absolutely necessary.

In the event of interference to the transmission, a coded audio signal, such as described hereinbefore, behaves very differently in dependence on which parts of the data frame have been interfered with.

In particular, interference with the header, bit allocation table and scaling factor selection information leads to large errors in the decoded signal. For this reason, an error protecting part or error protecting word (16 bit cyclic redundancy check, CRC) is frequently provided as an option, by means of which an error in the said region of the data frame can be detected and an appropriate protective measure can be initiated in the receiver (concealing the error, error concealment).

Somewhat amazingly, transmission errors within the coded audio sampling values in the audio data part only have a very small effect so that, even up to very high error rates (ca 10^{-2}), virtually no special error protection is required.

The object of the invention is to better protect the subband coded audio signals as described hereinbefore against transmission errors.

The object is achieved in accordance with the invention by a method of transmission having the features according to claim 1. Advantageous developments are specified in the appendant claims.

The invention is based on the perception that errors within the scaling factors - hereinafter referred to as scaling factor errors - have a very disturbing effect during reproduction. Thus, these scaling factor errors can sometimes lead to very loud "whistling" signals in the output of the receiver. Consequently, in order to prevent a user's hearing or loudspeaker membranes from being ruined, an effective supplementary error protection, or at least a detection of this scaling factor error, is necessary here in order that appropriate error concealment measures can be initiated.

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As one measure of protection in accordance with the invention, a supplementary item of error protecting information is inserted into the data frame for protecting the scaling factors. For this purpose, one (or more) CRC data words are preferably used. This then serves for detecting one or more transmission errors within the scaling factors.

Since it has been shown that a transmission error within the scaling factors only leads to a noticeably audible interference if one or more of the three MSBs (Most Significant Bit) of the scaling factors, which are coded with 6 bits, are interfered with, it is sufficient to monitor the three MSBs of the scaling factors by means of a CRC.

Further investigations have shown that it is not expedient to protect all of the scaling factors from all of the subbands together by means of a CRC but rather, to form e.g. four groups and thus, to monitor the scaling factors of only a certain number of subbands with the aid of a CRC. For example, for a total of 27 subbands, one can use the grading: 2, 4, 8, 13 or 4, 4, 8, 11 or 1, 1, 2, 4, 8, 11 or some other grading of the subbands. In addition, it has been shown for a coded stereo signal, that the scaling factors for the left and the right channels can be protected in common.

In order to be able to insert the supplementary CRC words in a compatible form into the known data frames, they are placed at the end of the data frame in the region of the meaningless filler bits. It is thereby ensured, with the help of the calculations for the bit allocation, that there always remains sufficient space for e.g. four CRC words each having e.g. 8 bits.

If there is no information relating to an audio signal present, then the supplementary CRC words are located at the end of the data frame, however, if such is present, then the supplementary CRC words are inserted directly in front of it.

The detection of transmission errors within the scaling factors can only occur if both the scaling factors themselves as well as the supplementary CRC words are available in the receiver. Consequently, as the supplementary CRC words are located practically at the end of the data frame, all of the data in the data frame has to be initially buffer stored before one can begin with the decoding. This can be avoided if the inserted CRC words in the currently progressing data stream relate to the scaling factors of the immediately succeeding data frame. This does of course lead to an increase in cost in the transmitter, something which is tolerable in a broadcasting system for example, but there is a distinct saving of storage space in the receiver.

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An embodiment of the invention will be explained in more detail hereinafter with the help of a drawing. In the drawing, illustrate

The further object is achieved in accordance with the invention by the characterising features of Claim 11. Herein, one starts from the premise that certain statistical features of an analogue audio signal are drawn upon for a continuous monitoring of the digitally coded, transmitted and decoded digital audio signal at the receiving end. One of the most important statistical features of natural audio signals consists in that rapid changes from low to high levels, but not however from high to low levels, occur in the level waveform taken with respect to time. A further feature of audio signals, that the alterations in the frequency response usually take place slowly and not abruptly, and, that they do not usually appear at the same time as large changes in level of the overall signal. This applies in particular for closely adjacent spectral components of the signal.

An embodiment of the invention will be explained in more detail hereinafter with the help of a drawing. In the drawing, there is illustrated in

Fig. 1a a frame format for the audio signal transmission

Fig. 1b a detailed breakdown of the frame format in accordance with Fig. 1a

Figs. 2a - d show waveforms with respect to time of scaling factors of a digitally coded audio signal that is transmitted without errors (Figs. 2a and 2b) as well as with errors (Figs. 2c and 2d).

Fig. 1a shows a general view of the frame format of a main frame which is transmitted as part of a bit stream of digitalised audio signals that has been divided up in this way in a frame like manner. The frame comprises a head part - a so-called header -, an error checking part - a so-called "Error Check" - , an audio data part and an ancillary data part - so-called "Ancillary Data Items" - relating to the audio signal. During the transmission, individual bits are interleaved (interleaving) within a frame or with the bits of another frame and are subjected to an optional scrambling (Scrambling).

Fig. 1b shows a detailed breakdown of the frame format in accordance with Fig. 1a. A 12 bit long synchronising word (all the bits having the value logical "1") is at the beginning of the header. Appended thereto is an identification bit ID,

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- a two bit item of "Layer" information regarding the data reduction algorithm that has been selected or the psycho-acoustic model of the coded audio signals that has been chosen,

- a protection bit including an item of information as to whether information theory redundancy is being transmitted,

- a bit rate index (four bit) including an item of information regarding the bit rate e.g. 32 kbit/sec or 128 kbit/sec that was selected for the coding,

- an item of information regarding the sampling frequency (two bit) e.g. 44.1 kHz that was selected for sampling the audio signals,

- a padding bit which is at logical "1" only for the sampling rate of 44.1 kHz,

- a private bit (not laid down)

- an item of information (two bit) regarding the selected mode e.g. stereo, intensity stereo, mono, etc.,

- an item of mode extension information (two bit) regarding the selected subbands for the intensity stereo mode,

- a copyright bit, wherein logical "1" means "protected by copyright",

- an original/home bit, including an item of information as to whether the transmitted audio signals originate from an original recording or live transmission or from a copy,

- an item of emphasis information (two bit).

The error checking part (Error Check) comprises a 16 bit CRC parity check word which is provided in particular for the detection of errors of transmission errors in the header, especially of the synchronising word and the scaling factor selection information and the bit allocation table in the audio data part.

In essence, the audio data part comprises a bit allocation table, items of scaling factor information, scaling factors, items of sample coding information, sampling values, meaningless filler bits and supplementary items of error protecting information at the end of the audio data part or of the data frame. The position of the filler bits and of the supplementary error protecting information may also be inter-exchanged as shown in Fig. 1b.

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The ancillary data part relating to the audio signal is defined by the user.

The supplementary error protecting information in the data frame, which is inserted directly at the end of the data frame before the ancillary data information relating to the audio signal, is especially relevant for the protection of data in the following data frame. Preferably, it only serves for the protection of the scaling factors transmitted in the following data frame. It is expedient thereby if the supplementary error protecting information serves for the protection of the MSBs of the scaling factors. If the number of scaling factors per subband is three, then too, only the three MSBs have to be protected.

Preferably, the supplementary error protecting information serves for the common protection of the scaling factors of the first and second channel of a stereo or two sound channel system. The supplementary error protecting information itself consists of one or more CRC words and replaces the filler bits provided at the positions in the data frame allocated thereto.

It is especially advantageous, if the supplementary error protecting information consists of four CRC words, which each protect the scaling factors of a number of subbands. Moreover, the invention relates to a method of the type previously described and for the detection of transmission errors on digital transmission paths for audio signals.

For the detection of transmission errors during the transmission of digital audio signals, it is known to add redundant bits e.g. filler bits, to the useful digital bit stream which permit transmission errors to be detected at the receiving location and, if necessary, to correct them (error protecting code). In many cases, the detection of transmission errors within the whole or within parts of the useful signal is not possible because either

- there is no error protection present,

- the error protection is insufficient,

- the error protection is related to the whole signal and interference with specific signal components cannot be particularly identified.

The object of the invention is to better protect the subband coded audio signals described hereinbefore against transmission errors. A further object of the present invention consists in improving the detection of transmission errors at the receiving end, in particular within specific signal components.

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The first object is achieved in accordance with the invention by a transmission method including the features in accordance with Claim 1. Advantageous developments are specified in the appendant Claims 2 to 10.

A floating point technique is often used for the digital coding of audio signals. Frequently, the displacements of the decimal point are only permitted to a limited extent, for example 500 times in one second. The value in front of the decimal point is then constant for a specific period of time e.g. for 2 msecs. One refers to this value as a scaling factor. For its part, it is digitally coded, usually provided with a high degree of error protection and transmitted as a signal component within the digital bit stream in the network.

One can only expect from the characteristics of the level waveform of an audio signal that,

scaling factor values usually only alter slowly,

rapid alterations only occur from low to high level values.

The following checking criteria can be derived therefrom:

1. Normal alterations in the waveform level (Fig. 1a), wherein the height of each illustrated column is indicative of the instantaneous level value which applies for a specific period of time of 2 msec for example.

2. Normal alterations in the waveform level with "normal" jumps (Fig. 1b).

3. Scaling factors having isolated occurrences of a level increase which has (have) probably been transmitted with an error (Fig. 1c), wherein the scaling factor subject to error is indicated by the hatching.

4. Scaling factors having isolated occurrences of a level decrease which has (have) probably been transmitted with an error (Fig. 1d), wherein the scaling factors subject to error are indicated by the hatching.

In order to detect a level jump which is subject to error in accordance with Fig. 1c or 1d, the jump in level is compared with a signal value that could be expected on statistical grounds whereby the waveform levels before and after the jump in level are used as the basis for the comparison with the statistically expected value. In the case of Figures 1c and 1d, this means that after the occurrence of the jump in level marked by the hatching, it is determined that a new jump in level immediately occurs, but

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in the opposite direction, immediately after the jump or after the jumps in level that are under observation. Only a change in level above a defined threshold is regarded as a jump in level.

A scaling factor detected or categorised as containing an error in this manner can be easily replaced by the probably correct scaling factor, namely, either by extrapolation (repetition of the preceding scaling factor) or by interpolation between the scaling factors assumed to be correct.

The error detecting process described hereinbefore can be carried out at the receiving end by simple means without any great expense since, basically, it only has to be determined whether or not, after a jump in level, a new oppositely directed jump in level occurs. By means of the method in accordance with the invention, storage media, e.g. optical storage media such as a compact disc or recordable optical storage media such as a magneto-optic disc or chip card stores, can now be used, in which the digitalised audio signals are stored, in which the digitalised audio signal is subjected to a subband coding, that the apparatus comprises a store for storing the received audio signal or selected components of this signal, as well as, level jump detecting means, which in the event of the occurrence of jumps in the level of the received audio signal or in selected components of this signal, compare the signal waveforms before and after the relevant jump in level with such signal waveforms as correspond to the statistics of the signal and that the jumps in level under observation are to be considered as errors if their value deviates from a signal waveform that could be expected on statistical grounds and which is stored in the store.

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Claims

1. Method of transmitting, storing and/or receiving digital audio signals in which the digitalised audio signals are subjected to a subband coding wherein one or more scaling factors are determined for each subband, the thus coded signals are transmitted in succeeding data frames and decoded and reproduced at the receiving end, wherein a data frame comprises a plurality of parts, namely

- a head part - a so-called header - including digitalised items of control information such as a synchronising word, an item of bit rate information, an item of sampling information, items of mode information etc,

- optionally, an error checking and/or correcting part preferably including a CRC protection word (Cyclic Redundancy Check),

- an audio part including amongst other things, a bit allocation table, an item of scaling factor selection information, one or more scaling factors per transmitted subband, sampling values, substantially meaningless filler bits and

an ancillary information part including items of ancillary information relating to the audio signals,

characterised in that, a supplementary item of error protecting information or an error protecting bit is inserted into the data frame for protecting the scaling factors.

2. Method in accordance with Claim 1, characterised in that, the supplementary item of error protecting information is inserted directly at the end of the data frame.

3. Method in accordance with Claim 1 or 2, characterised in that, the supplementary item of error protecting information is inserted directly before the item of ancillary information relating to the audio signals.

4. Method in accordance with any of the Claims 1 - 3, characterised in that, the supplementary item of error protecting information is associated with items of data in the succeeding data frame.

5. Method in accordance with any of the Claims 1 - 4, characterised in that, the supplementary item of error protecting information is associated with the 3 MSBs of the scaling factors.

6. Method in accordance with any of the Claims 1 - 5, characterised in that, the supplementary item of error protecting

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information is associated with the scaling factors of the first and second channels of a stereo or two sound channel signal.

7. Method in accordance with any of the Claims 1 - 6, characterised in that, the supplementary item of error protecting information consists of one or more CRC data words.

8. Method in accordance with any of the Claims 1 - 7, characterised in that, the supplementary item of error protecting information consists of four CRC data words which each protect the scaling factors of a number of subbands.

9. Method in accordance with any of the Claims 1 - 8, characterised in that, the data frame has a length of approximately 24 msecs.

10. Method in accordance with any of the Claims 1 - 9, characterised in that, meaningless filler bits are replaced by the supplementary item of error protecting information.

11. Method for the - a head part - a so-called header - including digitalised items of control information such as a synchronising word, an item of bit rate information, an item of sampling information, items of mode information etc,

- optionally, an error checking and/or correcting part preferably including a CRC protection word (Cyclic Redundancy Check),

- an audio part including amongst other things, a bit allocation table, an item of scaling factor selection information, one or more scaling factors per transmitted subband, sampling values, substantially meaningless filler bits and

an ancillary information part including ancillary items of information relating to the audio signals,

and furthermore, a method for the detection of transmission errors on digital transmission paths for the audio signals, characterised in that, in the event of the occurrence of jumps in the level of the received audio signal or in selected components of this signal e.g. the scaling factors, the signal waveforms before and after the relevant jump in level are compared with such signal waveforms as correspond to the statistics of the signal and that the jumps in level under observation are considered as errors if their value deviates from a signal waveform that could be statistically expected.

12. Apparatus for the reception and/or reproduction of a digital audio signal transmitted in accordance with the method of any of the Claims 1 - 10, characterised in that, the apparatus comprises

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a store for storing the supplementary item of error protecting information and means for error correcting and/or concealing faulty scaling factors, that the means is connected to the store and that the means uses the supplementary item of error protecting information for the scaling factors for the error correction and/or the concealment of the errors.

13. Apparatus for the reception and/or reproduction of a digitalised audio signal transmitted in accordance with the method of any of the Claims 1 - 11, characterised in that, the apparatus comprises a store for storing the received audio signal or selected components of this signal as well as level jump detecting means which, in the event of the occurrence of jumps in the level of the received audio signal or in selected components of this signal, compare the signal waveforms before and after the relevant jump in level with such signal waveforms as correspond to the statistics of the signal and that the jumps in level under observation can be considered as errors if their value deviates from a signal waveform that could be statistically expected and which is stored in the store.

14. Storage medium e.g. an optical or magnetic store, having stored digitalised audio signals in which the digitalised audio signals are subjected to a subband coding, wherein one or more scaling factors are determined for each subband, the thus coded signals are stored in succeeding data frames, wherein a data frame comprises a plurality of parts, namely

- a head part - a so-called header - including digitalised items of control information such as a synchronising word, an item of bit rate information, an item of sampling information, items of mode information etc,

- optionally, an error checking and/or correcting part preferably including a CRC protection word (Cyclic Redundancy Check),

- an audio part including amongst other things, a bit allocation table, an item of scaling factor selection information, one or more scaling factors per transmitted subband, sampling values, substantially meaningless filler bits and

an ancillary information part including items of ancillary information relating to the audio signals,

characterised in that, a supplementary item of error protecting information or an error protecting bit is inserted into the data frame for protecting the scaling factors.

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15. Storage medium in accordance with Claim 13, characterised in that, the supplementary item of error protecting information is inserted directly at the end of the data frame.

16. Storage medium in accordance with Claim 13 or 14, characterised in that, the supplementary item of error protecting information is inserted directly before the ancillary item of information relating to the audio signals.

17. Method in accordance with any of the Claims 13 - 15, characterised in that, the supplementary item of error protecting information are associated with data of the succeeding data frame.

18. Method in accordance with Claim 13 - 16, characterised in that, the supplementary item of error protecting information is associated with the three MFBs of the scaling factors.

19. Method in accordance with Claim 13 - 18, characterised in that, the supplementary item of error protecting information is associated in scaling factors in the first and second channels of a stereo or two sound channel signal.

20. Method in accordance with any of the Claims 13 - 19, characterised in that, the supplementary item of error protecting information consists of one or more CRC data words.

21. Storage medium having stored digitalised audio signals, characterised in that, the supplementary item of error protecting information consists of four CRC data words which each protect the scaling factors of a number of subbands.

22. Storage medium in accordance with any of the Claims 1 - 21, characterised in that, the data frame has a length of approximately 24 msecs.

23. Storage medium in accordance with any of the Claims 13 - 22, characterised in that, meaningless filler bits are replaced by the supplementary item of error protecting information.

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Claims

1. Method of transmitting, storing and/or receiving digital audio signals wherein one or more scaling factors is/are associated with subbands and the thus coded signals are transmitted in succeeding data frames and decoded and reproduced at the receiving end, wherein a data frame comprises the following parts:

- a header including digitalised items of control information such as e.g. a synchronising word, an item of bit rate information, an item of sampling information, items of mode information;

- an error checking and/or correcting part including a first error protection code,

- an audio data part
characterised by:
- the first error protection code is a CRC protection word and only relates to the header;
- the audio data part includes amongst other things, a bit allocation table, an item of scaling factor selection information, the scaling factors associated with the subbands, and sampling values;

- an ancillary information part including ancillary items of information relating to the audio signals, wherein a further CRC word or further CRC words for protecting the MSBs of the scaling factors is or are inserted at the end of the data frame in the region of the items of ancillary information relating to the audio signals.

2. Method in accordance with Claim 1, characterised in that, the further CRC protection words are associated with the MSBs of the scaling factors of the succeeding data frame.

3. Method in accordance with Claim 1 or 2, characterised in that, groups of scaling factors are formed within a data frame, a respective CRC protection word being associated with each group.

4. Method in accordance with one or more of the Claims 1 - 3, characterised in that, in the event of the occurrence of jumps in the components relating to the level e.g. the scaling factors, of the received audio signals, the signal waveforms before and after the relevant jump in level are compared with such signal waveforms as correspond to the statistics of the signal and that the jumps in level under observation are considered as errors and

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are corrected accordingly if their value deviates from a signal waveform that could be statistically expected.

5. Method in accordance with one or more of the Claims 1 - 4, characterised in that, the further CRC protection words are associated with the three MSBs of the scaling factors.

6. Method in accordance with one or more of the Claims 1 - 5, characterised in that, the further CRC protection words are associated in common with the scaling factors of the first and second channels of a stereo or two sound channel signal.

7. Method in accordance with one or more of the Claims 1 - 6, characterised in that, filler bits are occupied by the further CRC protection words.

8. Apparatus which receives a digital audio signal transmitted in accordance with the method of any of the Claims 1 - 7, characterised in that, the apparatus comprises a store for storing the further CRC protection words and means for error correcting and/or concealing faulty scaling factors, which means are connected to the store, and that the means use the further CRC protection words for the error correction and/or concealment for the scaling factors.

9. Apparatus in accordance with Claim 8, characterised in that, the apparatus comprises a store for storing the received audio signal or selected components of this signal, as well as, level jump detecting means which, in the event of the occurrence of jumps in the level of the received audio signal or in the selected components of the audio signal, compare the signal waveforms before and after the relevant jump in level with such signal waveforms as correspond to the statistics of the signal and that the jumps in level under observation are considered as errors and are corrected accordingly if their value deviates from a signal waveform that could be statistically expected and which is stored in the store.

10. Storage medium e.g. an optical or magnetic store, having stored digitalised audio signals which are coded by a method in accordance with one or more of the Claims 1 - 7.

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