

Description

FIELD OF THE INVENTION.

The invention relates to an encoding system, comprising a subband coder for subband coding of a wideband digital signal, for example, a digital audio signal, having a specific sampling frequency F_s , the subband coder including signal splitting means for generating, in response to the wideband digital signal, a plurality of M subband signals with sampling frequency reduction, for which purpose the splitting means split up the wideband digital signal into successive subbands having band numbers m which augment with frequency, where $1 \leq m \leq M$, which encoding system further comprises quantizing means for quantizing block-by-block the respective subband signals, a quantized subband signal comprising successive signal blocks, each signal block in a subband SB_m comprising q samples, the q samples in a signal block being represented each by n_m bits, the quantizing means including bit need determining means for determining a bit need b_m per signal block for corresponding signal blocks in the subbands, which bit need is related to the number of bits by which samples in a signal block in the subband SB_m should be represented, and the quantizing means including bit allocation means for allocating the available quantity of bits B to the various subbands in response to the need for bits determined by the bit need determining means so as to obtain the values n_m , where m ranges from 1 to M , and which encoding system further comprises accommodating means for accommodating the samples quantized with a certain number of bits in a signal block in a frame of a second digital signal having successive frames, the accommodating means further being arranged for accommodating the scale factor information in a frame, which information comprises x -bit words, an x -bit word representing a scale factor associated to the samples in a signal block, and the invention further relates to a transmitter comprising the encoding system and to an encoding method for subband coding the wideband digital signal.

BACKGROUND OF THE INVENTION

The encoding system of the type mentioned in the preamble is known from European Patent Application No. 289.080, prior-art document (1) in the list of documents stated at the end of the descriptions of the drawing Figures.

SUMMARY OF THE INVENTION

The invention specifically relates to the bit need determining means, determining the bit needs b_1 to b_M for corresponding signal blocks in the subbands SB_1 to SB_M on the basis of output signals of the subband

filters.

The invention therefore has for an object to derive the bit needs by implementing a novel method.

For this purpose, the encoding system according to the invention is characterised in that the bit need determining means are arranged for estimating the power v_m in a signal block in a subband SB_m for corresponding signal blocks in the subbands, for determining the sample SF_m having the maximum absolute value in the signal block,

- for calculating the magnitude w_m according to the formula

$$w_m = \sum_{i=1}^M d_{mi} v_i + W_{r,m}$$

- for calculating b_m according to the formula

$$b_m = K_1 \cdot 2 \log(K_2 \sqrt{SF_m^2 / w_m} + K_3)$$

in that d_{mi} is a matrix coefficient in an $M \times M$ matrix $[D]$, this matrix coefficient denoting the coefficient by which the power v_i in the subband i is multiplied to calculate the value of the masked power in the subband m as a result of the signal in the signal block in the subband i , in that $w_{r,m}$ is a measure for the masking threshold in the subband m and in that K_1 , K_2 and K_3 are constants.

It should be observed in this context that a bit need for the corresponding signal blocks in the various subbands has already been determined in the prior art. However, different algorithms have been used and different assumptions have been made there.

For example, in the prior-art document (3) a signal-to-noise ratio is determined per subband. This signal-to-noise ratio, expressed in dB, and divided by 6, then yields the bit need in a subband.

The coefficients K_1 and K_2 are preferably selected to be equal to 1 or $1/\sqrt{3}$. K_3 has a wider range of possibilities as this constant has less influence on the eventual result of the encoding. For example, one may take the value of 1 for K_3 , or omit K_3 altogether.

For the calculation of the bit needs a logarithmic representation for the various magnitudes is preferably used. This is advantageous in that a relatively small word length (number of bits by which the various magnitudes are represented) will be sufficient for the various magnitudes whereas, still a sufficient relative accuracy for the bit needs can be realised. This implies that the electronics for the realisation of the bit need determining means may be of simpler structure.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will be further explained in the following descriptions with reference to a number of exemplary embodiments, in which:

Fig. 1 shows an encoding system according to the invention;

Fig. 1a shows the corresponding signal blocks in the subband signals SB_1 to SB_M , each signal block comprising q samples;

Fig. 2 shows the quantization to a three-bit binary representation;

Fig. 3 shows the positions of the bit needs $b_1, b_2, \dots$, along the value axis;

Fig. 4 shows the method of determining the bit needs $b_1, \dots, b_M$;

Fig. 5 shows the method of allocating bits to the subbands;

Fig. 6 shows the initial bit allocation;

Fig. 7 shows the correction Table to be used for number additions utilizing a logarithmic representation of the numbers;

Fig. 8 shows a hardware design of the bit need determining means;

Fig. 9 shows a hardware design of the bit allocation means;

Fig. 10 shows the use in a transmitter in the form of a recording arrangement for recording quantized subband signals on a magnetic record carrier;

Figs. 11, 12 and 13 show the different allocation stages in dependence on the power value v_i , and Fig. 14 shows a diagram of the unit for generating the control signals featuring the different allocation stages.

DESCRIPTION OF (PREFERRED) EMBODIMENTS.

The description of the drawings will in essence relate to the description of the subband encoding of a single digital signal. That is to say, the subband encoding of a mono audio signal, or the subband encoding of only the right or rather the left signal portion of a stereo audio signal. This implies that in each subband signal there is only one subband signal present.

At the end of this description it will be explained how the bit need may be determined in case of subband encoding of a stereo signal, it being understood that there are two subband signals in each subband.

Fig. 1 shows the encoding system according to the invention. A wideband digital signal is applied to input terminal 1. An audio signal having a bandwidth of approximately 20 kHz may be considered in this respect.

Input 1 is supplied, for example, with 16-bit samples of the audio signal at a sampling frequency of 44 kHz. The audio signal is applied to a subband coder

2 which comprises signal splitting means. The subband coder 2 distributes the audio signal over M subbands by means of M filters, that is to say, a low-pass filter LP, $M-2$ band-pass filters BP and a high-pass filter HP. M is, for example, equal to 32. The sampling frequency of the M subband signals is reduced in the blocks referenced 9. In these blocks the sampling frequency is reduced by a factor M . The signals thus obtained are available at the outputs 3.1, 3.2, ..., 3. M . At the output 3.1 the signal is available in the lowest subband SB_1 . At the output 3.2 the signal is available in the lowest but one subband SB_2 . At the output 3. M the signal is available in the highest subband SB_M . The signals at the outputs 3.1 to 3. M have the structure of successive samples expressed in numbers of 16 bits or over, for example 24. In the exemplary embodiment of the invention under discussion the subbands SB_1 to SB_M all have the same width. This is not a necessity, however.

In the prior-art publication (5), by Krasner, a subdivision into a plurality of subbands is provided whose bandwidths approximately correspond to the bandwidths of the critical bands of the human auditory system in the respective frequency areas. The operation of the subband coder 2 will not further be explained since the operation of a subband coder has already been described extensively hereinbefore. For this purpose, reference is made to the prior-art documents (1), (5) and (7) which are assumed to be incorporated in this application where necessary.

The subband signals are combined to successive signal blocks of q successive samples, cf. Fig. 1a, and are applied to corresponding quantizers Q_1 to Q_M . In a quantizer Q_m the samples are quantized to quantized samples having a number of bits n_m smaller than 16.

Fig. 2 shows a quantization with a 3-bit binary representation. During the process of quantization the signal blocks (groups) of q successive samples of the subband signals are each time quantized with a number of bits (3) in the example of Fig. 2. For example, q is equal to 12. In addition, the q samples in a signal block are normalized first. This normalization is performed by dividing the amplitudes of the q samples by the amplitude of the sample having the largest absolute value in the signal block. The amplitude of the sample having the largest amplitude in the signal block of subband SB_m yields the scale factor SF_m , cf. documents (2a), (2b). Subsequently, the amplitudes of the normalized samples, which are now situated in an amplitude range from -1 to +1, are quantized according to Fig. 2.

This implies that samples in the amplitude range between -1 and -0.71 are quantized with the 3-bit number 000, samples in the amplitude range from -0.71 to 0.42 are quantized with 001, samples in the amplitude range from 0.42 to 0.14 are quantized with 010, samples in the amplitude range from -0.14 to

0.14 are quantized with 001, samples in the amplitude range from 0.14 to 0.42 are quantized with 100, samples in the amplitude range from 0.42 to 0.71 are quantized with 101 and samples in the amplitude range from 0.71 to 1.00 are quantized with 110.

In the prior-art document (2b) this three-bit quantization is extensively discussed, cf. Figs. 24, 25 and 26 and the relevant descriptions in that document.

The quantized samples in the subbands SB_1 to SB_M are thereafter available at the respective outputs 4.1 to 4.M.

Furthermore, the outputs 3.1 to 3.M are coupled to the respective inputs 5.1 to 5.M of bit need determining means 6. The bit need determining means 6 determine the bit need b_m for the time equivalent signal blocks of q samples in the subbands SB_1 to SB_M . The bit need b_m is a number related to the number of bits with which the q samples in a q -sample signal block in a subband signal should be quantized.

The bit needs b_1 to b_M , derived by the bit need determining means 6, are applied to bit allocation means 7. The bit allocation means 7 determines the actual number of bits n_1 to n_M with which the q samples of the corresponding signal blocks in the subband signals SB_1 to SB_M are quantized. Control signals corresponding to the numbers n_1 to n_M are applied to the respective quantizers Q_1 to Q_M through lines 8.1 to 8.M, so that the quantizers can quantize the samples with the correct number of bits.

The following will provide an explanation of the operation of the bit need determining means 6 and the bit allocation means 7. The bit needs for the time-equivalent signal blocks of q samples in the subband signals SB_1 to SB_M are derived from estimates for the power v_m and the scale factor SF_m in a signal block in a subband signal SB_m .

The power v_m may, for example, be obtained by means of the following formula:

$$v_m = \frac{1}{q} \sum_{i=1}^q s_i^2$$

where s_i is the amplitude of the i^{th} sample in the q -sample signal block in the subband SB_m . The scale factor SF_m was equal to the amplitude of the sample in the signal block having the largest absolute value, as has already been observed hereinbefore.

It should be observed in this context that the estimate for the power v_m in a signal block in the subband SB_m also might have been obtained by assuming that v_m was equal to the squared scale factor SF_m .

For all corresponding signal blocks in the subband signals SB_1 to SB_M the power v_m and the scale factor SF_m are determined in this fashion. The powers are ordered in a vector $\{v\}$. By multiplying the vector $\{v\}$ by an $M \times M$ matrix $[D]$, one will obtain a vector $\{w\}$

utilizing the following formula:

$$\{w\} = [D] \{v\} + \{w_r\}$$

In this formula $[D]$ is a matrix whose coefficients d_{ij} denote the coefficient by which the power v_i of the q -sample signal block in the subband signal SB_i is to be multiplied to calculate the masked power in the subband SB_i due to the signal in the q -sample signal block in the subband signal SB_j , and $w_{r,i}$ is the coefficient in the vector $\{w_r\}$ which denotes the masking threshold in the subband SB_i . Thus, $w_{r,m}$ bears a relation to the maximum signal power of the signal in a subband SB_m which signal is not audible.

The vector $\{w\}$ now has the coefficients w_i which are estimates for the masked quantizing noise in each subband SB_i . Quantizing-noise in subband SB_i which has a power of less than w_i is thus inaudible. The coefficients d_{ij} of the matrix $[D]$ are to be calculated according to the literature, cf. prior-art document (4). The bit need b_1 to b_M can be calculated with the aid of the following formula:

$$b_m = K_1 \cdot 2 \log(\sqrt{SF_m^2/3w_m} + 1)$$

more generally the following may hold:

$$b_m = K_1 \cdot 2 \log(K_2 \sqrt{SF_m^2/3w_m} + K_3)$$

The former formula can simply be derived from the latter formula by assuming: $K_2 = 1/\sqrt{3}$ and $K_3 = 1$; K_1 , K_2 and K_3 are constants, for which it holds that K_1 is preferably equal to approximately 1 and K_2 is preferably equal to approximately $1/\sqrt{3}$. K_3 has a wider range of possibilities. It may be assumed that K_3 will be less than 10, where K_3 will, for example, preferably be taken to be equal to 1 or may be neglected. In addition, in the latter case there will be a simpler implementation of the calculation.

The coefficients (bit needs) b_1 to b_M obtained in this fashion, are situated in a specific amplitude range. The coefficients may be negative and non-integers. A coefficient b_m bears a relation to the number of bits with which the samples in the q -sample signal block of a subband signal SB_m should be quantized, so that it holds that if b_{m1} for subband signal SB_{m1} is greater than b_{m2} for the subband signal SB_{m2} , the number of bits with which the q samples in a signal block in the subband signal SB_{m1} should be quantized will have to be greater than the number of bits with which the q samples of a time-equivalent signal block in the subband signal SB_{m2} should be quantized.

This is shown in qualitative terms with reference to Fig. 3. In Fig. 3 seven bit needs b_1 to b_6 , $b_{\max}$ and $b_{\min}$ are plotted along the value axis. $b_{\max}$ is the bit need having the maximum value and $b_{\min}$ is the bit need having the minimum value. It will be noticed that $b_{\min}$, b_2 and b_6 are negative and that, furthermore, the following holds: $b_{\min} < b_6 < b_2 < b_4 < b_3 < b_{\max}$. In terms of quality it may now be assumed that the subband signal SB_{m1} , with $b_m = b_{\min}$, should/could be quantized with the minimum number of bits and the subband signal SB_{m2} , with $b_m = b_{\max}$, with the maximum number

of bits.

Fig. 4 shows a flow chart of the operation of the bit need determining means 6.

Fig. 4 shows the program for determining the bit needs b_1 to b_M for time-equivalent q -sample signal blocks in the subband signals SB_1 to SB_M . In this case only a single q -sample signal block in a subband signal is concerned. For a successive q -sample signal block in the subband signal, and the signal blocks of the other subband signals corresponding with the signal block (in case of a time-parallel subband signal supply), the operation represented in Fig. 4 will be performed once again.

The operation starts at block 10. First, the running variable m is set to 1 (block 12). Then, the q samples $S_1, \dots, S_q$ of a signal block in a subband signal SB_m are inputted (block 14) and the power V_m is calculated (block 16). Also the scale factor SF_m (block 18) is determined.

The blocks 14, 16 and 18 are repeated for all subband signals via the loop through blocks 20 and 22. If the values V_m and SF_m have been determined for all corresponding signal blocks, the matrix calculation will be performed in order to obtain the vector $\{w\}$ (block 24).

Subsequently, m is again set to 1 (block 26) and the bit need (b_m) is determined (block 28) for all subbands via the loop through the blocks 30 and 32 after which the operation is terminated (block 34). The bit needs b_m are determined in block 28 in accordance with the formula stated previously in which the constants K_1 , K_2 and K_3 are taken to be equal to the values of 1, $1/\sqrt{3}$ and 0 respectively.

The method of Fig. 4 shows the time-consecutive calculations of the coefficients v_m in the vector $\{v\}$, compare the loop through block 22 in the program, and the method shows the time-consecutive calculations of the bit needs b_1 to b_M , compare the loop through block 32. This is a very suitable method, more specifically, if the corresponding signal blocks having the samples s_1 to s_q for the successive subbands SB_1 , $SB_2, \dots, SB_{M-1}$, SB_M , are applied serially.

If the signal blocks are applied in parallel, the calculation of the coefficients v_m could be performed in parallel for all subbands and thus the loop through block 22 is avoided. Likewise, the bit needs b_1 to b_M may be calculated in parallel and this will render the loop through block 32 redundant.

The operation of the bit allocation means 7 will now be explained. The flow chart of Fig. 5 will be used for this purpose. The program determines for time-equivalent q -sample signal blocks in the subband signals SB_1 to SB_M the values n_1 to n_M from the bit needs b_1 to b_M . Here too it is a matter of a single signal block having q samples in a subband signal. For a directly successive q -sample signal block in the subband signal and the time-equivalent signal blocks in the other subband signals, the method of Fig. 5 will be carried

out again.

It is now assumed that, after quantization, B_0 bits are available for transmitting the overall information connected with the M signal blocks of q samples of, for example, 24 bits each. Assuming that after quantization, R bits are available per sample, averaged over the subbands, it holds that B_0 is equal to the largest integer smaller than $M \cdot q \cdot R$.

In the prior-art documents (2a) and (2b) it is shown that not only the quantized samples are transmitted, but also the scale factors SF_1 to SF_M (scale factor information) and the bit allocation information (that is to say, information which bears a relation to the number of bits with which the samples in a signal block in a subband signal are quantized, i.e. the values n_1 to n_M). The bit allocation information is then represented by $y=4$ bits for each n_m . Thus, this implies that, actually, only $B = B_0 - y \cdot M$ bits are available for the transmission of M signal blocks of quantized subband signals and the scale factor information.

The documents (2a) and (2b) further describe that the y -bit number ($y=4$) 0000 in the bit allocation information denotes that no samples are transmitted in the relevant subband. In that case no scale factor information for this subband will be transmitted either. The scale factor information for a subband is represented by means of an x -bit number ($x=6$).

The method of bit allocation is now as follows. The method starts at block 40, Fig. 5. Subsequently, all numbers n_m are first set to zero. Then an initial bit allocation is performed at block 44. This initial bit allocation will be explained in the following with reference to Fig. 5. Thereafter, the maximum bit need is determined. This is the bit need b_j . In the example of Fig. 3 this would be b_{max} . Then it is considered whether n_j is greater than or equal to a certain value n_{max} (block 48). In the present example n_{max} is equal to 16. This means that the quantized samples can only be represented by binary numbers with a maximum of 16 bits.

If n_j is greater than or equal to n_{max} , the q -sample signal block in the subband j will be excluded from the allocation of further bits. For this purpose, the bit need b_j is made equal to a so-called "flag value" (block 66). The flag value is represented in Fig. 3 and is a value smaller than the minimum bit need b_{min} . If c_1 in the block 56 to be discussed hereinbelow is greater than unity, n_j might be greater than n_{max} . In addition, n_j will then be assumed to be equal to n_{max} at block 66.

If n_j is equal to zero (block 50), the program will proceed through the blocks 52 and 54. At block 54, a_1 bits are initially allocated to the subband SB_j . This means $n_j = a_1$. The total number B of available bits now decreases by $a_1 \cdot q \cdot x$. The q quantized samples of the signal block in the subband signal SB_j are represented each by a_1 bits and, in addition, an x -bit-long scale factor SF_j is to be added. Furthermore, in block

54 the bit need b_j is decreased by a value a_2 . If n_j is unequal to zero, the program will proceed through block 56. The number of bits n_j is now increased by c_1 . The total number B of available bits now decreases by $c_1 \cdot q$, due to the fact that the q quantized samples of a signal block are now represented by an additional number of c_1 bits.

Naturally, bit allocation only takes place if there are still sufficient bits available. Therefore, block 52 is present. Because, if there are insufficient bits available the program will proceed through block 66 at which the relevant bit need b_j is again made equal to the flag value. The signal block in the subband concerned is then excluded from further bit allocation.

As long as there are bit needs that have values greater than the flag value (block 58) and as long as there are still sufficient bits available (block 60), the program will return through circuit 62 to block 46 for a next calculation of the maximum bit need. If all bit needs b_m are smaller than or equal to the flag value, the program will stop. The program will also stop if there are insufficient bits to be allocated (block 60).

The method is characterised in that when a first bit allocation is performed (block 54), the number of allocated bits (a_1) is greater than the number of bits of one or more subsequent allocations (block 56) (c_1), worded differently $a_1 > c_1$. Furthermore, it holds that a_2 is greater than or equal to unity. Preferably, a_1 is equal to a_2 and c_1 equal to c_2 . a_1 , a_2 , c_1 and c_2 are numbers greater than zero. a_1 and c_1 are preferably integers. But, this is not a necessity. An example may be shown for this purpose.

It is assumed that one wishes to quantize the q samples in a signal block in five levels. For this purpose, 3 bits are needed per sample. However, this is not an efficient encoding because a subdivision into seven levels of 3 bits is possible.

If, however, three samples are combined, these three samples with five signal level will present 125 options. These 125 options may be represented by means of a 7-bit binary number. Thus, no more than 7/3 bits per sample. n_m would in that case be equal to 7/3. This will provide a more efficient encoding.

It has been indicated hereinbefore that when the quantized samples are transmitted, both scale factor information and bit allocation information are co-transmitted. The scale factor information then has the form of x -bit words, of which each x -bit word denotes a scale factor SF_m that belongs to the q samples in a signal block in the subband SB_m . The bit allocation information then has the form of y -bit words, of which each y -bit word denotes a number of bits n_m by which each sample in a signal block in the subband SB_m is represented. Previously, the prior-art documents (2a) and (2b) have already been referred to.

If in the bit need determining means 6 only the scale factors SF_m are used for calculating the powers v_m , that is to say, because v_m is assumed to be equal

to the square of SF_m , the bit allocation information need not be co-transmitted.

On the receiver side the bit needs b_1 to b_M can be derived from the transmitted scale factors SF_m and on the basis of these needs the magnitudes n_1 to n_M , while implementing the calculation method as discussed hereinbefore. Thus, the receiver likewise comprises bit need determining means which derive the powers v_m from the scale factors SF_m and derive from these powers the bit needs b_m , and also includes bit allocation means which is capable of calculating the magnitudes n_1 to n_M on the basis of the bit needs b_m and the available bit quantity which in this case is equal to B_0 . Since, as observed hereinbefore, $B=B_0 \cdot y \cdot M$, the latter method is advantageous in that more bits can be allocated to the subbands.

It may sometimes be necessary to pre-allocate a number of bits to a signal block in a subband SB_m , for example, in the case where there are signal blocks which must not be quantized with zero bits irrespective of their bit need. The reason for this is that signal blocks must not be switched on or off in an unqualified manner. This would produce audible effects.

The above also shows that it may sometimes be necessary or useful to exclude a signal block in advance from bit allocation. For this purpose, block 44 in the program of Fig. 5 is inserted. Fig. 6 shows an elaboration of block 44. Two signal blocks are represented there, that is to say, the signal blocks in the subbands SB_k and SB_l to which have been pre-allocated a number of bits A_{k0} or A_{l0} . This implies that $n_k = A_{k0}$ and $n_l = A_{l0}$. From the bit needs b_k and b_l the respective values A_{k1} and A_{l1} , are subtracted and the remaining number of bits B is reduced by $A_{k0} \cdot q \cdot x$ and $A_{l0} \cdot q \cdot x$ respectively. Actually, for A_{k0} and A_{l0} the same holds as for a_1 . Preferably, $A_{k0} = A_{l0} = a_1$. For A_{k1} and A_{l1} the same holds as for a_2 . Preferably, $A_{k1} = A_{l1} = a_2$.

To the signal blocks in the subbands SB_k and SB_l may naturally be allocated more bits, as required, at block 56 of the method presented in Fig. 5.

Furthermore, at block 44 of Fig. 6 it is shown that the signal block in the subband SB_l is excluded from bit allocation. For this purpose, the bit need b_l for this signal block is made equal to the flag value.

Figs. 11, 12 and 13 represent the situations in which there is initial bit allocation or no initial bit allocation to the subbands. The Figures show the consecutive time intervals ΔT in which a group of M corresponding signal blocks of the M subbands are processed. In each time interval the power $v_i(t)$ and the magnitude $w_i(t)$ are determined for each subband SB_i . If $v_i(t)$ is greater than $w_i(t)$, there will be initial bit allocation to the subband SB_i . As will be evident from Fig. 11, this holds for periods of time situated before $t=t_1$. Fig. 14 shows a circuit by means of which, on the basis of the magnitudes v_i and w_i , control signals may be derived which denote whether initial bit allocation

is to take place, in which case the output of the SR flip-flop 140 is "high" or "logic 1", or whether no bit allocation is to take place, in which case the output of the SR flip-flop 141 is "high", or whether no initial bit allocation takes place, in which case the output of a counter 142 is "high". In the latter case bits may still be allocated to the subband in question, but the allocation then takes place at block 54 and may also take place at block 56 according to the method of Fig. 5. These control signals may thus be applied to block 44 in Fig. 6 and denote what functions are to be performed in this block.

At the instant $t=t_1$, $v_i(t)$ becomes smaller than $w_i(t)$. The output 144 of the comparator 143 now becomes "low" whereas the output 145 of this comparator becomes "high". Through the OR-gate 147 this "high" signal is applied to the AND-gate 148 so that clock pulses are passed to the AND-gate 149 at a rate f equal to $1/\Delta T$. Since a "high" signal is applied to the other input of the AND-gate 149 through the inverter 150, the clock pulses are passed to the input 151. The counter 142 now counts down from the initial position 5 (decimal) under the influence of the clock pulses, cf. Fig. 11. Since the output of the counter 142 remains "low", the position of the flip-flop 140 does not change so that the initial bit allocation is maintained.

One time interval later, $v_i(t)$ is again larger than $w_i(t)$. The output 144 of the comparator 143 becomes "high" again which implies that the rising edge is fed to the set input of the counter 142 through the OR-gate 152. The count of counter 142 is reset to 5 (decimal). At the instant t_2 , cf. Fig. 11, $v_i(t)$ again becomes smaller than $w_i(t)$. Now, $v_i(t)$ remains smaller than $w_i(t)$ for a sufficiently long period of time to make it possible for the counter 142 to count back until the 0 count (decimal) is reached. This is reached at the instant $t=t_3$, cf. Fig. 11. At that moment the output of the counter 142 becomes "high". The flip-flop 140 is reset. Through inverter 150 and AND-gate 149 the counting operation of counter 142 is blocked so that it retains the 0 count.

Initial bits are no longer allocated to this subband. At the instant $t=t_4$, $v_i(t)$ again becomes larger than $w_i(t)$. The counter 142 is reset to the count 5 and, in addition, the flip-flop 140 is set in a manner so that initial bits are again allocated.

Fig. 12a represents a situation in which $v_i(t)$, prior to the instant at which the counter 142 is reset to Zero, already becomes smaller than a specific threshold value v_{thr} . At the instant $t=t_5$ the output 145 of the comparator 143 becomes "low" again and the output 146 "high". Since the inverter 153 applies a "high" signal to one input of the AND-gate 154, the "high" signal is conveyed to the AND-gate 148 through the AND-gate 154 and the OR-gate 147. The counter 142 continues to count. The phase of initial bit allocation is thus maintained until the count 0 (decimal) is reached. The output of the counter 142 now briefly rises. This en-

tails that flip-flop 141 is set through the AND-gate 155. Through the AND-gate 156 and the OR-gate 152 the high output signal of the flip-flop 141 is applied to the set input of the counter 142, which immediately afterwards jumps to count 5 (decimal). In addition, the further down-counting of the counter 142 is blocked because the inverter 153 applies a "low" signal to the one input of the AND-gate 154. From instant t_6 onwards there is no bit allocation whatsoever to the relevant subband.

Fig. 12b represents the situation in which $v_i(t)$ has remained in the range between v_{thr} and $w_i(t)$ sufficiently long so that the phase of "no initial bit allocation" has commenced. At the instant t_7 , v_i will be smaller than v_{thr} . At that moment the output 145 will become "low" and the output 146 "high".

At that moment the flip-flop 141 is set through the AND-gate 155 and the counter 142 is reset to the count 5 through the AND-gate 156 and the OR-gate 152. The output of the counter 142 thus becomes "low" and the output of the flip-flop 141 "high". There is no bit allocation.

Fig. 13 shows a situation in which $v_i(t)$ increases again. At the instant t_8 , $v_i(t)$ becomes larger than v_{thr} . The output 145 becomes "high" so that the counter 142 may count down. One time interval later $v_i(t)$ is again smaller than v_{thr} . The output 146 becomes "high" again so that the counter is reset to the count of 5 through the AND-gate 156 and the OR-gate 152. If $v_i(t)$ is greater than v_{thr} for a sufficiently long period of time, the counter 142 can count down to zero. At $t=t_9$, the output of counter 142 becomes "high". Through the AND-gate 159, to which a "high" signal is applied through the inverter 158, the flip-flop 141 is reset, so that at this moment the phase of "no initial bit allocation" is terminated and changed into the phase of "no initial bit allocation".

In the following a simplified calculation of the bit need b_m will be explained. In this calculation a logarithmic representation is used for the various magnitudes which play a part in the calculation. This is possible because not absolute but relative precision of the bit needs is concerned with respect to the calculation of the bit needs b_1 to b_M .

In logarithmic representation a number g is approximated by $g=r^k$, where r is a fixed base greater than unity and the power k is selected to be an integer. The number g is approximated in the best possible way by a right choice of k . The integer k is used as a representation of g . In the calculation for the bit need b_m there are both multiplications of two numbers and additions of two numbers. Multiplications in the logarithmic representation correspond with the addition of the powers. That is to say, if $g_1=r^{k_1}$ and $g_2=r^{k_2}$, the logarithmic representation of $g_1 \cdot g_2$ will be equal to k_1+k_2 .

For the logarithmic representation of additions of these two numbers g_1 and g_2 , the following holds. As-

suming that $g_1 > g_2$, it holds that $g_1 + g_2 = r^{k_1} + T(k_1 - k_2)$. The logarithmic representation for $g_1 + g_2$ is thus equal to $k_1 + T(k_1 - k_2)$. $T(k_1 - k_2)$ is a correction factor in the form of an integer which may be derived from a Table. Fig. 7 shows a Table of this type for $r = 2^{1/16}$. The value for r equalling $2^{1/16}$ can be obtained from an accuracy analysis of the bit needs b_m .

It may further be inferred that the calculation of the needs b_m in a logarithmic representation with a properly selected base r , in lieu of the customary calculations in a linear fixed-point representation, considerably reduces the word widths of the numbers. In addition, no multiplier-accumulator is necessary for calculating the vector $\{w\}$, but only a simple accumulator and a Table having a restricted number of entries. The Table of Fig. 7, for example, narrows down to a ROM having contents smaller than 0.5 kbit. The series of numbers stored in the ROM is relatively small. In addition, these numbers are arranged in a specific order. Therefore, it is possible to reduce the look-up Table even more at the cost of some logic.

It should be observed in this context that the logarithmic representation for the addition of two numbers as described hereinbefore is known *per se* by the name of Zech logarithm and described in document (6), compare, more specifically, chapter 3, section 4, page 191.

The bit need determining means 6 and the bit allocation means 7 may be arranged in software. However, hardware designs are also possible. For example, Fig. 8 shows a hardware design of the bit need determining means 6.

Fig. 8 shows the corresponding signal blocks in the subband signals SB_1 to SB_M , which are serially applied to input 70. The first sample s_1 of the subband SB_1 is applied first and the last sample s_q of the subband SB_M is applied last.

In the largest sample determining means 71 the largest sample SF_m is determined for each signal block, which value is then stored in a memory 72. In a squaring unit 73 the samples are squared and thereafter applied to an input of an adder 74. The output of the adder 74 is coupled to an input of a memory 75. The output of that memory 75 is coupled both to a second input of the adder 74 and an input of a divider 76. The elements referenced 74, 75 and 76 determine the magnitude v_m for each signal block, compare block 16 in Fig. 4. To this end, the first sample s_1 of a signal block 3 is squared into the subband signal SB_m in the squaring unit 73 and, in adder 74, added to the value stored in memory 75, which value is momentarily zero, and thereafter stored in the memory 75. Subsequently, the second sample s_2 is squared, added to the value stored in memory 75 and then stored in that memory. This is continued until the last sample s_q is squared and added to the value stored in memory 75. The sum thus obtained in the memory 75 is equal to

$$\sum_{i=1}^q s_i^2$$

which then, after a division by q in the divider 76, is stored as a coefficient v_m in memory 77. Similar calculations are made for the corresponding signal blocks of the further subbands until all coefficients of the vector $\{v\}$ have been stored in the memory 77. The bit need determining means 6 further include a memory 78 for storing the matrix coefficient d_{m1} of the matrix $[D]$ and a memory 79 for storing the coefficients $w_{r,m}$ of the vector $\{w_r\}$. Outputs of the memories 77 and 78 are coupled to inputs of the multiplier 80. An output of the multiplier 80 is coupled to a first input of an adder 81 whose one output is coupled to the input of a memory 82. The output of the memory 82 is coupled both to a second input of the adder 81 and to a first input of an adder 83.

The elements referenced 80, 81 and 82 are intended to perform the matrix multiplication $[D]\{v\}$.

During this operation the value d_{m1} from memory 78 is multiplied by the value v_1 from memory 77 for a specific subband m and the result is added by adder 81 to the value present in memory 82 at that moment, which is zero, and then stored in memory 82. After this, d_{m2} is multiplied by v_2 and the result is added to the value stored in memory 82. This is continued until d_{mM} is multiplied by v_M and the result is added to the value stored in memory 82. At that moment the value

$$\sum_{i=1}^M d_{mi} v_i$$

is stored in memory 82. In adder 83, the value $w_{r,m}$ stored in memory 79 is added to this result. The value w_m thus obtained is stored in memory 84. This procedure is reiterated for the corresponding signal blocks in the further subbands until all coefficients of the vector $\{w\}$ are stored in memory 84.

Subsequently, for each subband SB_m , the magnitudes SF_m and w_m are read from the memories 72 and 84 and applied to the calculation unit which eventually determines the bit need b_m . This bit need is stored in a memory 86. This calculation is also performed for the further subbands until all bit needs b_1 to b_M are stored in the memory 86.

The procedure may be reiterated for a successive series of M signal blocks. Also the arrangement shown in Fig. 8 utilises the fact that there is serial information supply. If the signal blocks were supplied in parallel, the calculation could largely be performed in parallel. This means, for example, that the circuit

comprising the elements 71, 73, 74, 75 and 76 could occur M times in the arrangement. The circuit comprising elements 80, 81, 82 and 83 could then likewise occur M times.

Fig. 9 shows a hardware embodiment of the bit allocation means 7. The bit allocation means comprises a memory 90 in which the number of bits B still to be allocated have been stored, a memory 91 in which the values n_1 to n_M are stored and a memory 92 in which the bit needs b_1 to b_M have been stored. This memory 92 could correspond with memory 86 of Fig. 8. At the beginning of an allocation cycle the initial value for B is stored in memory 90, which value is available at terminal 94. Furthermore, the initial values for the bit needs b_m have been stored in memory 92 whereas memory 91 stores all zeros fed to terminal 93 by means of reset signals.

Subsequently, detector 95 determines the maximum value of the bit needs stored in memory 92. This may, for example, be realised by successively reading out all bit needs b_1 to b_M at the output 96 and applying these bit needs through line 97 to the input 98 of the detector 95. At the output 99, the detector 95 provides the index of the maximum bit need b_j . This index j is used as an address for addressing through line 100 the locations in the memories 91 and 92 in which the values are stored for n_j , b_j respectively, so that these values are available at the respective outputs 101 and 96. Output 101 is coupled to an input of an $n_j=0$ detector 102. If the detector 102 detects $n_j=0$, it provides at output 103 a control signal which is applied to control signal inputs of controllable switches S_1 , S_2 and S_3 . These switches then assume different positions from the ones shown in the drawing Figure. This implies that in a subtractor 105 the value $a_{1,q+k}$ is subtracted from the value B available at output 106 of memory 90, and this new value is again applied to input 107 of this memory through line 104 so that the new value is stored in memory 90. Furthermore, through lines 108 and 109 the value a_1 which is available at terminal 110, is applied to the input 111 of memory 91. Switch S_4 then has the position shown in the drawing Figure. This value a_1 is stored in memory 91 as a new value for n_j . In the subtractor 112 the value a_2 is subtracted from the value b_j available at the output 96 of memory 92. The value thus obtained is applied to input 115 through lines 113 and 114, while the switch S_5 has the position shown in the drawing Figure, so that the new value for b_j can be stored in the memory location b_j in memory 92.

The method described hereinbefore is the method as denoted in block 54 in Fig. 5.

If the detector 102 detects that n_j is unequal to zero, no (or a different) control signal is generated. Switches S_1 , S_2 and S_3 then have the positions shown in the Figure. The value $c_{1,q}$ is now subtracted from the value B stored in memory 90 and the result thus obtained is stored again in memory 90. In the adder

117 the value c_1 is added to the value n_j which is read from the memory 91 through output 101. Again through lines 108 and 109 the new value for n_j is applied to input 111 of memory 91 to be stored in the memory 91.

Furthermore, in subtractor 112 the value c_2 is subtracted from the value b_j present at the output 91, and the value thus obtained is applied to input 115 through lines 113 and 114 in order to be stored in memory 92. The method described actually corresponds with block 56 of the method shown in Fig. 5.

In the method of Fig. 5 there is further shown that the decision block 48, $n_j \geq n_{\max}$?, provides that b_j is made equal to the flag value (block 66 in Fig. 5), and n_j equal to $n_{\max}$ (should it turn out to be necessary). In the circuit of Fig. 9 this has been taken into account by means of the $n_j \geq n_{\max}$ detector 118. If detector 118 detects a situation in which $n_j \geq n_{\max}$, it generates at its output 119 a control signal which is applied to the control input of the controllable switch S_4 and, through an OR-gate 120, to the control input of the controllable switch S_5 , which switches then assumes the positions different from the ones shown in the Figure. The value $n_{\max}$, applied to the terminal 121, is now applied to the input 111 of memory 91. $n_{\max}$ is then stored in the memory location for n_j in memory 91. Accordingly, the flag value, block 122, is applied to input 115 so that the flag value is stored in the memory location for b_j in memory 92.

It will be evident that there is a central control unit (not shown) which detects the output signal of detector 118 and, on detection of this signal, applies only load pulses to the memories 91 and 92 for storing there $n_{\max}$ and the flag value. No load pulse is then applied to memory 90, since the value B in the memory is to remain unchanged.

Furthermore, the flag value is allocated to b_j if both n_j is equal to zero and $B \geq a_{1,q+x}$, compare the blocks 50, 52 and 66 in Fig. 5. Thereto the circuit of Fig. 9 includes the detector 123 and the AND-gate 124. At the occurrence of detection signals of both detector 103 and detector 123, switch S_5 is again set to the other position than the position shown and the flag value b_F is stored in memory 92 in location j. In this case the central processor will generate a load pulse only for memory 92 and no load pulses for memories 90 and 91.

It will be self-evident that the initial bit allocation as described with reference to Fig. 6 may also be implemented here, for example, controlled by the necessary control and address signals from the central controller. This will not be explained any further because after the above explanation it may be assumed to be within the grasp of those skilled in the art.

Fig. 10 shows the use of the subband coder as described hereinbefore, in a transmitter, especially a transmitter in the form of a recording arrangement for recording the quantized subband signals in one or

more tracks on a magnetic record carrier.

The section referenced 130 is the subband coder discussed hereinbefore, which produces the quantized subband signals at the outputs 4.1 to 4.M.

The section referenced 131 converts these signals into a second digital signal which is available at output 132. This second digital signal comprises successive frames of which the format is extensively discussed in the prior-art documents (2a) and (2b). Also the structure of the block 131 is explained in these documents.

The section referenced 133 renders the second digital signal suitable for recording on a record carrier, for example, a magnetic record carrier 134. The unit 133 thereto comprises an 8-to-10 converter. In a converter of this type 8-bit data words in the serial information stream are converted to 10-bit code words. Furthermore, interleaving may take place. All this has for its object to enable error correction of the received information on the receiver side (when reproducing the record carrier).

The output signal of block 133 is applied to recording means 135 by which the signal is recorded in one or more longitudinal tracks on the record carrier 134. The recording means 135 thereto comprise one or more recording heads 136.

For a further explanation of the arrangement of Fig. 10, the prior-art document (8) be referred to, which is also assumed to be incorporated where necessary in this Patent Application.

It should further be observed that the invention is not restricted to the depicted exemplary embodiments only. Various modifications of the embodiments described are feasible without leaving the scope of the invention as defined in the claims.

In the foregoing the bit need determination and bit allocation have been described for a number of M subband signals, while there was always a single subband signal (for example, a mono signal) in each subband.

However, the invention may also be applied to a subband coder for subband encoding of a stereo signal. This implies that there are two subband signals in each subband, that is to say, a left and a right subband signal.

Two optional ways of subband encoding of a stereo signal will be briefly discussed hereinbelow. A first option is to process the left and right subband signals separately in the manner described above. The M subband signals SB_1 to SB_M as discussed above are then, for example, the M left subband signals. The procedure discussed hereinbefore is then carried out for these left subband signals. In the bit need determining means 6 first the bit needs b_{11} to b_{M1} are determined. Thereafter, the numbers of bits to be allocated, i.e. n_{11} to n_{M1} , are determined in the bit allocation means 7. In the procedure discussed above and explained with reference to Fig. 5, the val-

ue B was used for the bit allocation. B was then equal to the number of available bits. It may be obvious that in the present case just half this number of available bits B is used for determining n_{11} to n_{M1} . The other half of the number of available bits will then be used for bit allocation to the right subband signals.

The arrangement for the stereo signal subband encoding according to the first option actually comprises twice the arrangement shown in Fig. 1. The second section of the arrangement thus comprises a subband splitter, such as the splitter 2, for generating the M right subband signals. Furthermore, a bit need determining means is present, such as unit 6, which determines the bit needs b_{1r} to b_{Mr} , and a bit allocation means, such as unit 7, which derives therefrom the numbers of allocated bits n_{1r} to n_{Mr} . Also for this purpose, half the actual number of available bits is available.

According to a second option for subband encoding of a stereo signal the bit needs b_{11} to b_{M1} and b_{1r} to b_{Mr} are derived in the same manner as in the first option. In contradistinction to the first option, in which the bit allocation for the left and right subband signals was performed separately, in the second option the 2M bit needs b_{11} to b_{M1} and b_{1r} to b_{Mr} are applied to a bit allocation unit such as unit 7, which then naturally has 2M inputs. In this unit the 2M numbers n_{11} to n_{M1} and n_{1r} to n_{Mr} are derived in a manner similar to the manner described with reference to Fig. 5 on the basis of the actually available number of bits. For this purpose the bit allocation means have 2M outputs.

It should further be observed that when a stereo signal is encoded, 2M values for the bit allocation information are concerned, represented each by y bits. This means that for the bit allocation procedure for a stereo signal no more than $B=B_0 \cdot 2 \cdot y \cdot M$ bits are available.

PRIOR ART:

- (1) European Patent Application No. 289.080 (PHN 12.108)
- (2a) Dutch Patent Application No. 89.01.401 (PHN 12.967)
- (2b) Dutch Patent Application No. 90.00.338 (PHN 13.241)
- (3) EBU Techn. Review, No. 230, August 1988 G. Theile et. al. "Low bit rate coding of high-quality audio signals. An introduction to the MASCAM system".
- (4) Philips Journal of Research '44, 329-343, 1989 R.N.J. Veldhuis et al. "Subband coding of digital audio signals".
- (5) IEEE ICASSP 80, Vol. 1, pp. 327-331, April 9-11, 1980 M.A. Krasner "The critical band coder ... Digital encoding of speech signals based on the perceptual requirements of the auditory system".
- (6) F.J. MacWilliams et al., "The theory of error

correcting codes", North Holland publishing comp. 1983

(7) European Patent Application No. 89201408.5 (PHQ 89.018)

(8) Dutch Patent Application No. 90.00.635 (PHN 13.281)

Claims

1. Encoding system, comprising a subband coder (2) for subband coding of a wideband digital signal, for example, a digital audio signal, having a specific sampling frequency F_s , the subband coder including signal splitting means for generating, in response to the wideband digital signal, a plurality of M subband signals with sampling frequency reduction, for which purpose the splitting means split up the wideband digital signal into successive subbands ($SB_1 \dots SB_M$) having band numbers m which augment with frequency, where $1 \leq m \leq M$, which encoding system further comprises quantizing means ($Q_1 \dots Q_M$, 6, 7) for quantizing block-by-block the respective subband signals, a quantized subband signal comprising successive signal blocks, each signal block in a subband SB_m comprising q samples, the q samples in a signal block being represented each by n_m bits, the quantizing means ($Q_1 \dots Q_M$) including bit need determining means (6) for determining a bit need b_m per signal block for corresponding signal blocks in the subbands, which bit need is related to the number of bits by which samples in a signal block in the subband SB_m should be represented, and the quantizing means including bit allocation means (7) for allocating the available quantity of bits B to the various subbands in response to the bit need determined by the bit need determining means (6) so as to obtain the values n_m , where m ranges from 1 to M , and which encoding system further comprises accommodating means for accommodating the samples quantized with a certain number of bits in a signal block in a frame of a second digital signal having successive frames the accommodating means further being arranged for accommodating the scale factor information in a frame, which information comprises x -bit words, an x -bit word representing a scale factor associated to the samples in a signal block, characterised in that the bit need determining means (6) are arranged for estimating the power v_m in a signal block in a subband SB_m for corresponding signal blocks in the subbands, for determining the scale factor SF_m corresponding to the maximum absolute value in the signal block,

- for calculating the magnitude w_m according to the formula

$$w_m = \sum_{i=1}^M d_{mi} v_i + W_{r,m}$$

- for calculating b_m according to the formula

$$b_m = K_1 \cdot 2 \log(K_2 \sqrt{SF_m^2 / w_m} + K_3)$$

in that d_{mi} is a matrix coefficient in an $M \times M$ matrix $[D]$, this matrix coefficient denoting the coefficient by which the power v_i in the subband i is multiplied to calculate the value of the masked power in the subband m as a result of the signal in the signal block in the subband i , in that $w_{r,m}$ is a measure for the masking threshold in the subband SB_m and in that K_1 , K_2 and K_3 are constants.

2. Encoding system as claimed in Claim 1, characterised in that for the calculation of w_m according to the formula

$$w_m = \sum_{i=1}^M d_{mi} v_i + W_{r,m}$$

the bit need determining means utilize a logarithmic representation for the magnitudes d_{mi} , v_i , w_m and $w_{r,m}$.

3. Encoding system as claimed in Claim 2, characterised in that the bit need determining means to this end comprise means for adding and multiplying numbers given in a logarithmic representation.
4. Encoding system as claimed in Claim 1, 2 or 3, characterised in that $K_2 = 1/\sqrt{3}$ and K_3 is preferably equal to either 1 or zero.
5. Encoding system as claimed in Claim 4, characterised in that $K_1 = 1$.
6. Encoding system as claimed in one of the Claims 1 to 5, characterised in that the bit need determining means are arranged for estimating for corresponding signal blocks in the subbands the power v_m in a signal block in a subband SB_m according to the formula:

$$v_m = \frac{1}{q} \sum_{i=1}^q s_i^2$$

where s_i are the amplitudes of the q samples in a



correcting codes", North Holland publishing comp. 1983

(7) European Patent Application No. 89201408.5 (PHQ 89.018)

(8) Dutch Patent Application No. 90.00.635 (PHN 13.281)

Claims

1. Encoding system, comprising a subband coder (2) for subband coding of a wideband digital signal, for example, a digital audio signal, having a specific sampling frequency F_s , the subband coder including signal splitting means for generating, in response to the wideband digital signal, a plurality of M subband signals with sampling frequency reduction, for which purpose the splitting means split up the wideband digital signal into successive subbands ($SB_1 \dots SB_M$) having band numbers m which augment with frequency, where $1 \leq m \leq M$, which encoding system further comprises quantizing means ($Q_1 \dots Q_M$, 6, 7) for quantizing block-by-block the respective subband signals, a quantized subband signal comprising successive signal blocks, each signal block in a subband SB_m comprising q samples, the q samples in a signal block being represented each by n_m bits, the quantizing means ($Q_1 \dots Q_M$) including bit need determining means (6) for determining a bit need b_m per signal block for corresponding signal blocks in the subbands, which bit need is related to the number of bits by which samples in a signal block in the subband SB_m should be represented, and the quantizing means including bit allocation means (7) for allocating the available quantity of bits B to the various subbands in response to the bit need determined by the bit need determining means (6) so as to obtain the values n_m , where m ranges from 1 to M , and which encoding system further comprises accommodating means for accommodating the samples quantized with a certain number of bits in a signal block in a frame of a second digital signal having successive frames the accommodating means further being arranged for accommodating the scale factor information in a frame, which information comprises x -bit words, an x -bit word representing a scale factor associated to the samples in a signal block, characterised in that the bit need determining means (6) are arranged for estimating the power v_m in a signal block in a subband SB_m for corresponding signal blocks in the subbands, for determining the scale factor SF_m corresponding to the maximum absolute value in the signal block,

- for calculating the magnitude w_m according to the formula

$$w_m = \sum_{i=1}^M d_{mi} v_i + W_{r,m}$$

- for calculating b_m according to the formula

$$b_m = K_1 \cdot 2 \log(K_2 \sqrt{SF_m^2 / w_m} + K_3)$$

in that d_{mi} is a matrix coefficient in an $M \times M$ matrix $[D]$, this matrix coefficient denoting the coefficient by which the power v_i in the subband i is multiplied to calculate the value of the masked power in the subband m as a result of the signal in the signal block in the subband i , in that $w_{r,m}$ is a measure for the masking threshold in the subband SB_m and in that K_1 , K_2 and K_3 are constants.

2. Encoding system as claimed in Claim 1, characterised in that for the calculation of w_m according to the formula

$$w_m = \sum_{i=1}^M d_{mi} v_i + W_{r,m}$$

the bit need determining means utilize a logarithmic representation for the magnitudes d_{mi} , v_i , w_m and $w_{r,m}$.

3. Encoding system as claimed in Claim 2, characterised in that the bit need determining means to this end comprise means for adding and multiplying numbers given in a logarithmic representation.
4. Encoding system as claimed in Claim 1, 2 or 3, characterised in that $K_2 = 1/\sqrt{3}$ and K_3 is preferably equal to either 1 or zero.
5. Encoding system as claimed in Claim 4, characterised in that $K_1 = 1$.

6. Encoding system as claimed in one of the Claims 1 to 5, characterised in that the bit need determining means are arranged for estimating for corresponding signal blocks in the subbands the power v_m in a signal block in a subband SB_m according to the formula:

$$v_m = \frac{1}{q} \sum_{i=1}^q s_i^2$$

where s_i are the amplitudes of the q samples in a



signal block of the subband SB_m .

7. Transmitter comprising the encoding system as claimed in one of the Claims 1 to 6.
8. Transmitter as claimed in Claim 7, characterised in that the transmitter has the form of a recording means for recording the quantized subband signals in a track on a record carrier.
9. Transmitter as claimed in Claim 8, characterised in that the record carrier is a magnetic record carrier.
10. Encoding method, for subband coding of a wideband digital signal, for example, a digital audio signal, having a specific sampling frequency F_s , in which the wideband digital signal is splitted into a plurality of M subband signals with sampling frequency reduction, for which purpose the wideband digital signal is splitted into successive subbands having band numbers m which augment with frequency, where $1 \leq m \leq M$, respective subband signals are quantized block-by-block, a quantized subband signal comprising successive signal blocks, each signal block in a subband SB_m comprising q samples, the q samples in a signal block being represented each by n_m bits, bit needs b_m are determined per signal block for corresponding signal blocks in the subbands, which bit need is related to the number of bits by which samples in a signal block in the subband SB_m should be represented, and an available quantity of bits B are allocated to the samples in corresponding signal blocks in the various subbands in response to the bit need determined so as to obtain the values n_m , where m ranges from 1 to M , and in which the samples quantized with a certain number of bits are accommodated in a frame of a second digital signal having successive frames, further scale factor information comprising x -bit words, an x -bit word representing a scale factor associated to the samples in a signal block is accommodated in said frame, characterized in that, for deriving bitneeds b_m , the method further comprises the steps of
 - estimating the power v_m (16) in a signal block in a subband SB_m for corresponding signal blocks in the subbands,
 - determining the scale factor SF_m (18) which corresponds to the maximum absolute value in the signal block,
 - calculating the magnitude w_m (24) according to the formula

$$w_m = \sum_{i=1}^M d_{mi} v_i + w_{r,m}$$

- for calculating b_m (28) according to the formula

$$b_m = K_1 \cdot 2 \log(K_2 \sqrt{SF_m^2 / w_m} + K_3)$$

in that d_{mi} is a matrix coefficient in an $M \times M$ matrix $[D]$, this matrix coefficient denoting the coefficient by which the power v_i in the subband is multiplied to calculate the value of the masked power in the subband m as a result of the signal in the signal block in the subband i , in that $w_{r,m}$ is a measure for the masking threshold in the subband SB_m and in that K_1 , K_2 and K_3 are constants.

Patentansprüche

1. Codieranordnung mit einem Unterbandcoder (2) zum Unterbandcodieren eines digitalen Breitbandsignals, beispielsweise eines digitalen Audiosignals, mit einer bestimmten Abtastrate F_s , wobei der Unterbandcoder mit Spaltmitteln versehen ist zum in Antwort auf das digitale Breitbandsignal Erzeugen einer Anzahl von M Unterbandsignalen mit Abtastatenverringern, wozu die Spaltmittel das digitale Breitbandsignal in aufeinanderfolgenden Unterbänder ($SB_1 \dots SB_M$) mit Bandnummern m aufteilen, die mit der Frequenz zunehmen, wobei für m gilt $1 \leq m \leq M$, und wobei die Codieranordnung weiterhin Quantisierungsmittel ($Q_1 \dots Q_M$, 6, 7) aufweist zum blockweisen Quantisieren der betreffenden Unterbandsignale, wobei ein quantisiertes Unterbandsignal aus aufeinanderfolgenden Signalblöcken aufgebaut ist, wobei jeder Signalblock in einem Unterband SB_m q Abtastwerte aufweist, wobei die q Abtastwerte in einem Signalblock mit je n_m Bits dargestellt sind, wobei die Quantisierungsmittel ($Q_1 \dots Q_M$) Bitbedarfbestimmungsmittel (6) aufweisen zum für entsprechende Signalblöcke in den Unterbändern je Signalblock Bestimmen eines Bitbedarfs b_m , der in einem Verhältnis zu der Anzahl Bits steht, mit der Abtastwerte in einem Signalblock in dem Unterband SB_m dargestellt werden müßten und mit Bitzuordnungsmitteln (7) zum Zuordnen der verfügbaren Bitmenge B zu den jeweiligen Unterbändern, je nach dem Bitbedarf, wie dieser durch die Bitbedarfbestimmungsmittel (6) bestimmt worden ist zum Erhalten der Werte n_m , wobei m von 1 bis einschließlich M geht, wobei die Codieranordnung weiterhin Mittel aufweist zum Unterbringen der mit einer bestimmten Menge Bits quantisierten Abtastwerte in einem Si-



gnalblock in einem Rahmen eines zweiten digitalen Signals mit aufeinanderfolgenden Rahmen, wobei die Mittel weiterhin dazu eingerichtet sind, in einem Rahmen Unterbringen von Skalenfaktorinformation, wobei die Skalenfaktorinformation x-Bits-Worte aufweist, wobei ein x-Bit-Wort einen den Abtastwerten in einem Signalblock zugeordneten Skalenfaktor darstellt, und auf einen Sender mit der Codieranordnung, sowie auf ein Codierverfahren für Unterbandcodierung des digitalen Breitbandsignals, dadurch gekennzeichnet, daß die Bitbedarfbestimmungsmittel dazu eingerichtet sind, für entsprechende Signalblöcke in den Unterbändern die Leistung v_m in einem Signalblock in einem Unterband SB_m zu schätzen, den Abtastwert SF_m mit dem größten Absolutwert in dem Signalblock zu bestimmen,

- die Größe w_m nach der untenstehenden Formel zu berechnen

$$w_m = \sum_{i=1}^M d_{mi} v_i + W_{r,m}$$

- den Wert b_m nach der untenstehenden Formel zu berechnen

$$b_m = K_1 \cdot 2 \log(K_2 \sqrt{SF_m^2 w_m} + K_3)$$

daß d_{mi} ein Matrixkoeffizient ist in einer $M \times M$ -Matrix $[D]$, wobei dieser Matrixkoeffizient den Multiplikationsfaktor angibt, mit dem die Leistung v_i in dem Unterband i multipliziert wird um zu berechnen, welche die maskierte Leistung in dem Unterband m ist infolge des Signals in dem Signalblock in dem Unterband 1, wobei $w_{r,m}$ ein Maß für die Maskierungsschwelle in dem Unterband SB_m ist und wobei K_1 , K_2 und K_3 Konstanten sind.

2. Codieranordnung nach Anspruch 1, dadurch gekennzeichnet, daß zur Berechnung von w_m nach der Formel

$$w_m = \sum_{i=1}^M d_{mi} v_i + W_{r,m}$$

die Bitbedarfbestimmungsmittel eine logarithmische Darstellung für die Größen d_{mi} , v_i , w_m und $w_{r,m}$ benutzen.

3. Codieranordnung nach Anspruch 2, dadurch gekennzeichnet, daß die Bitbedarfbestimmungsmittel dazu Mittel aufweisen zum Addieren und Multiplizieren von Zahlen in logarithmischer Dar-

stellung.

4. Codieranordnung nach Anspruch 1, 2 oder 3, dadurch gekennzeichnet, daß $K_2 = 1/\sqrt{3}$ und K_3 vorzugsweise gleich 1 oder Null ist.

5. Codieranordnung nach Anspruch 4, dadurch gekennzeichnet, daß $K_1 = 1$ ist.

6. Codieranordnung nach einem der Ansprüche 1 bis 5, dadurch gekennzeichnet, daß die Bitbedarfbestimmungsmittel dazu vorgesehen sind, für entsprechende Signalblöcke in den Unterbändern die Leistung v_m in einem Signalblock in einem Unterband SB_m nach der untenstehenden Formel zu schätzen:

$$v_m = \frac{1}{q} \sum_{i=1}^q s_i^2$$

wobei s_i die Amplituden der q Abtastwerte in einem Signalblock des Unterbandes sind.

7. Sender mit der Codieranordnung nach einem oder mehreren der Ansprüche 1 bis 6.

8. Sender nach Anspruch 7, dadurch gekennzeichnet, daß der Sender die Form eines Aufzeichnungsmittels zum Aufzeichnen der quantisierten Unterbandsignale in einer Spur auf einem Aufzeichnungsträger hat.

9. Sender nach Anspruch 8, dadurch gekennzeichnet, daß der Aufzeichnungsträger ein magnetischer Aufzeichnungsträger ist.

10. Codierungsverfahren für Unterbandcodierung eines digitalen Breitbandsignals, beispielsweise eines digitalen Audio-Signals, mit einer spezifischen Abtastrate F_s , wobei das digitale Breitbandsignal eine Anzahl von M Unterbandsignalen mit Abtastratenverringern aufgeteilt wird, wozu das digitale Breitbandsignal in aufeinanderfolgende Unterbänder mit Bandnummern m aufgeteilt wird, die mit der Frequenz zunehmen, wobei für m gilt $1 \leq m \leq M$, und wobei die betreffenden Unterbandsignale blockweise quantisiert werden, wobei ein quantisiertes Unterbandsignal aus aufeinanderfolgenden Signalblöcken aufgebaut ist,

wobei jeder Signalblock in einem Unterband SB_m q Abtastwerte aufweist, wobei die q Abtastwerte in einem Signalblock mit je n_m Bits dargestellt sind, wobei für entsprechende Signalblöcke in den Unterbändern diese Bitbedürfnisse b_m , der in einem Verhältnis zu der Anzahl Bits stehen, mit



der Abtastwerte in einem Signalblock in dem Unterband SB_m dargestellt werden müßten und mit Bitzuordnungsmitteln zum Zuordnen der verfügbaren Bitmenge B zu den jeweiligen Unterbändern, je nach dem Bitbedarf, wie dieser durch die Bitbedarfbestimmungsmittel bestimmt worden ist zum Erhalten der Werte n_m , wobei m von 1 bis einschließlich M geht, und wobei die mit einer bestimmten Menge Bits quantisierten Abtastwerte in einem Signalblock in einem Rahmen eines zweiten digitalen Signals mit aufeinanderfolgenden Rahmen untergebracht sind, wobei weiterhin Skalenfaktorinformation vorhanden ist, wobei die Skalenfaktorinformation x -Bits-Worte aufweist, wobei ein x -Bit-Wort einen den Abtastwerten in einem Signalblock zugeordneten Skalenfaktor darstellt, dadurch gekennzeichnet, daß zum Herleiten von Bitbedürfnissen b_m das verfahren weiterhin die nachfolgenden verfahrensschritte aufweist:

- das Schätzen der Leistung v_m (16) in einem Signalblock in einem Unterband SB_m für entsprechende Blöcke in den Unterbändern,
- das Bestimmen des Skalenfaktors SF_m (18), der dem maximalen Absolutwert in dem Signalblock entspricht,
- das Berechnen der Größe w_m (24) nach der Formel:

$$w_m = \sum_{i=1}^M d_{mi} v_i + W_{r,m}$$

- das Berechnen des Wertes b_m (28) nach der untenstehenden Formel

$$b_m = K_1 \cdot 2 \log(K_2 \sqrt{SF_m^2 / w_m} + K_3)$$

daß d_{mi} ein Matrixkoeffizient ist in einer $M \times M$ -Matrix $[D]$, wobei dieser Matrixkoeffizient den Multiplikationsfaktor angibt, mit dem die Leistung v_i in dem Unterband i multipliziert wird um zu berechnen, welche die maskierte Leistung in dem Unterband m ist infolge des Signals in dem Signalblock in dem Unterband 1, wobei $w_{r,m}$ ein Maß für die Maskierungsschwelle in dem Unterband m ist und wobei K_1 , K_2 und K_3 Konstanten sind.

Revendications

1. Système de codage comprenant un codeur de sous-bandes (2) pour codage en sous-bandes d'un signal numérique à large bande, par exemple, un signal audio numérique, ayant une fré-

quence d'échantillonnage spécifique F_s , le codeur de sous-bandes comprenant des moyens de division de signaux pour générer, en réaction au signal numérique à large bande, une pluralité de M signaux de sous-bandes avec une réduction de la fréquence d'échantillonnage, les moyens de division divisant à cet effet le signal numérique à large bande en des sous-bandes successives ($SB_1 \dots SB_M$) ayant des nombres de bandes m qui augmentent avec la fréquence, où $1 \leq m \leq M$, ledit système de codage comprenant, en outre, des moyens de quantification ($Q_1 \dots Q_M$, 6, 7) pour quantifier bloc par bloc les signaux de sous-bandes respectifs, un signal de sous-bande quantifié comprenant des blocs de signaux successifs, chaque bloc de signal d'une sous-bande SB_m comprenant q échantillons, les q échantillons d'un bloc de signal étant représentés chacun par n_m bits, les moyens de quantification ($Q_1 \dots Q_M$) comprenant des moyens de détermination des besoins en bits (6) pour déterminer un besoin en bits b_m par bloc de signal pour le bloc de signal correspondant dans les sous-bandes, ledit besoin en bits étant en rapport avec le nombre de bits par lequel des échantillons d'un bloc de signal de la sous-bande SB_m doivent être représentés, et les moyens de quantification comprenant des moyens d'affectation de bits (7) destinés à affecter la quantité disponible de bits B aux diverses sous-bandes en réaction au besoin en bits déterminé par les moyens de détermination des besoins en bits (6) afin d'obtenir des valeurs n_m , où m se situe dans une plage de 1 à M , ledit système de codage comprenant, en outre, des moyens d'accueil pour recevoir les échantillons quantifiés avec un certain nombre de bits dans un bloc de signal dans une trame d'un deuxième signal numérique ayant des trames successives, les moyens d'accueil étant, en outre, à même de recevoir les informations de facteur d'échelle dans une trame, ces informations comprenant des mots de x bits, un mot de x bits représentant un facteur d'échelle associé aux échantillons dans un bloc de signal, caractérisé en ce que les moyens de détermination des besoins en bits (6) sont agencés pour estimer la puissance v_m d'un bloc de signal dans une sous-bande SB_m pour des blocs de signaux correspondants des sous-bandes, pour déterminer le facteur d'échelle SF_m correspondant à la valeur absolue maximum dans le bloc de signal,

- pour calculer la grandeur w_m selon la formule :

$$w_m = \sum_{i=1}^M d_{mi} v_i + W_{r,m}$$



- pour calculer la grandeur b_m selon la formule :

$$b_m = K_1 \cdot 2 \log(K_2 \sqrt{SF_m^2 / w_m} + K_3)$$

en ce que d_{mi} est un coefficient de matrice dans une matrice $M \times M$ [D], ce coefficient de matrice désignant le coefficient par lequel la puissance v_i de la sous-bande i est multipliée pour calculer la valeur de la puissance masquée dans la sous-bande m suite au signal dans le bloc de signal de la sous-bande i , en ce que $w_{r,m}$ est une mesure du seuil de masquage de la sous-bande SB_m et en ce que K_1 , K_2 et K_3 sont des constantes.

2. Système de codage selon la revendication 1, caractérisé en ce que, pour le calcul de w_m selon la formule :

$$w_m = \sum_{i=1}^M d_{mi} v_i + w_{r,m}$$

les moyens de détermination des besoins en bits utilisent une représentation logarithmique pour les grandeurs d_{mi} , v_i , w_m et $w_{r,m}$.

3. Système de codage selon la revendication 2, caractérisé en ce que les moyens de détermination des besoins en bits comprennent à cette fin des moyens pour additionner et multiplier des nombres donnés dans une représentation logarithmique.
4. Système de codage selon la revendication 1, 2 ou 3, caractérisé en ce que $K_2 = 1/\sqrt{3}$ et K_3 est de préférence égal à 1 ou à zéro.
5. Système de codage selon la revendication 4, caractérisé en ce que $K_1 = 1$.
6. Système de codage selon l'une quelconque des revendications 1 à 5, caractérisé en ce que les moyens de détermination des besoins en bits sont agencés pour estimer, pour les blocs de signaux correspondants dans les sous-bandes, la puissance v_m d'un bloc de signal dans une sous-bande SB_m selon la formule :

$$v_m = \frac{1}{q} \sum_{i=1}^q s_i^2$$

où s_i est l'amplitude des q échantillons dans un bloc de signal de la sous-bande SB_m .

7. Emetteur comprenant le système de codage selon l'une quelconque des revendications 1 à 6.

8. Emetteur selon la revendication 7, caractérisé en ce que l'émetteur se présente sous la forme d'un moyen d'enregistrement pour enregistrer les signaux de sous-bandes quantifiés sur une piste d'un support d'enregistrement.

9. Emetteur selon la revendication 8, caractérisé en ce que le support d'enregistrement est un support d'enregistrement magnétique.

10. Procédé de codage pour le codage en sous-bandes d'un signal numérique à large bande, par exemple, un signal audionumérique, ayant une fréquence d'échantillonnage spécifique F_s , dans lequel le signal numérique à large bande est divisé en une pluralité de M signaux de sous-bandes avec une réduction de la fréquence d'échantillonnage, le signal numérique à large bande étant divisé à cet effet en des sous-bandes successives ayant des nombres de bandes m qui augmentent avec la fréquence, où $1 \leq m \leq M$, des signaux de sous-bandes respectifs sont quantifiés bloc par bloc, un signal de sous-bande quantifié comprenant des blocs de signaux successifs, chaque bloc de signal dans une sous-bande SB_m comprenant q échantillons, les q échantillons d'un bloc de signal étant représentés chacun par n_m bits, des besoins en bits b_m sont déterminés par bloc de signal pour des blocs de signaux correspondants dans les sous-bandes, ledit besoin en bits étant en rapport avec le nombre de bits par lequel des échantillons d'un bloc de signal de la sous-bande SB_m doivent être représentés, et une quantité disponible de bits B est affectée aux échantillons de blocs de signaux correspondants dans les diverses sous-bandes en réaction au besoin en bits déterminé afin d'obtenir les valeurs n_m , où m se situe dans une plage de 1 à M , et dans lequel les échantillons quantifiés avec un certain nombre de bits sont reçus dans une trame d'un deuxième signal numérique ayant des trames successives, d'autres informations de facteurs d'échelle comprenant des mots de x bits, un mot de x bits représentant un facteur d'échelle associé aux échantillons dans un bloc de signal, sont reçues dans ladite trame, caractérisé en ce que, pour obtenir les besoins en bits b_m , le procédé consiste à :

- estimer la puissance v_m (16) d'un bloc de signal dans une sous-bande SB_m pour des blocs de signaux correspondants dans les sous-bandes,
- déterminer le facteur d'échelle SF_m (18) qui correspond à la valeur absolue maximum dans le bloc de signal,



- calculer la grandeur w_m (24) selon la formule :

$$w_m = \sum_{i=1}^M d_{mi} v_i + W_{r,m} \quad 5$$

- calculer la grandeur b_m (28) selon la formule :

$$b_m = K_1 \cdot 2 \log(K_2 \sqrt{SF_m^2 / w_m} + K_3) \quad 10$$

en ce que d_{mi} est un coefficient de matrice dans une matrice $M \times M$ [D], ce coefficient de matrice désignant le coefficient par lequel la puissance v_i de la sous-bande i est multipliée pour calculer la valeur de la puissance masquée dans la sous-bande m suite au signal dans le bloc de signal de la sous-bande i , en ce que $w_{r,m}$ est une mesure du seuil de masquage de la sous-bande SB_m et en ce que K_1 , K_2 et K_3 sont des constantes. 15

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